Adobe® Audition® Help

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Chapter 1: Introduction to Adobe Audition

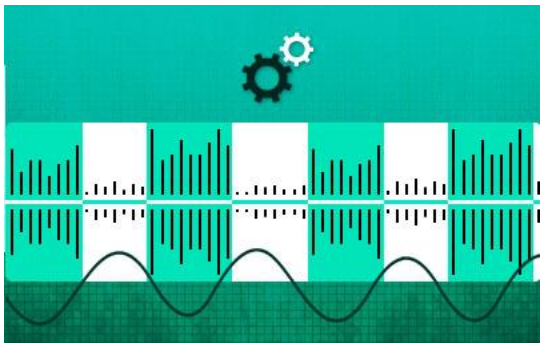
What's new in Adobe Audition

Note:

The version 13.0 and version 12. x releases of Adobe audition provides stability and performance improvements.

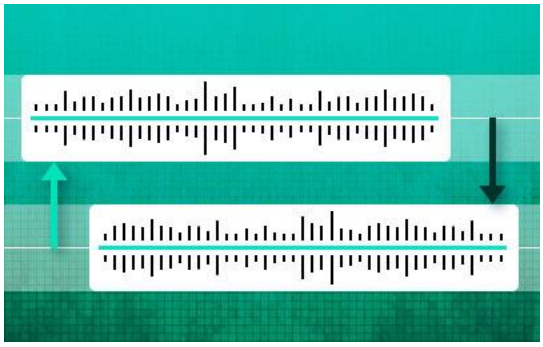
Top new features

November 2019 (version 13.0)



Improved effects support

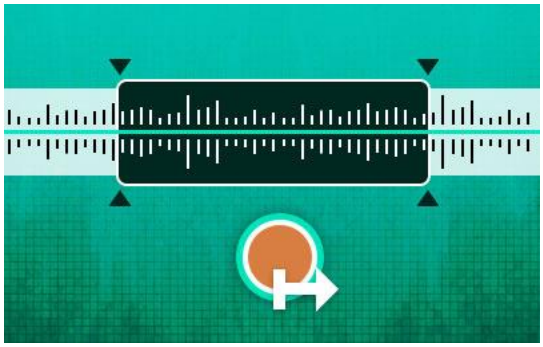
Route multichannel effects channels with full control.



Overlapped clips support

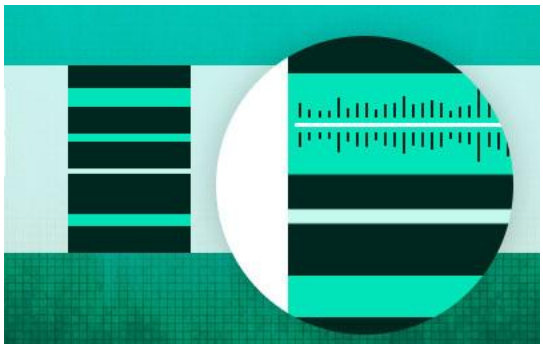
Audition now allows you to play or mute overlapping clips simultaneously.

April 2019 release (version 12.1)



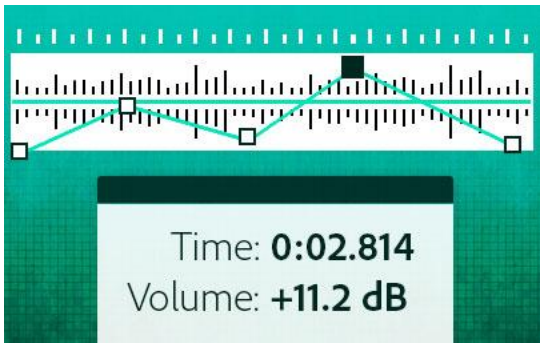
Punch and roll recording

Easily interact with voiceover recording by punching into portions of the take and rolling up to and through the re-take.



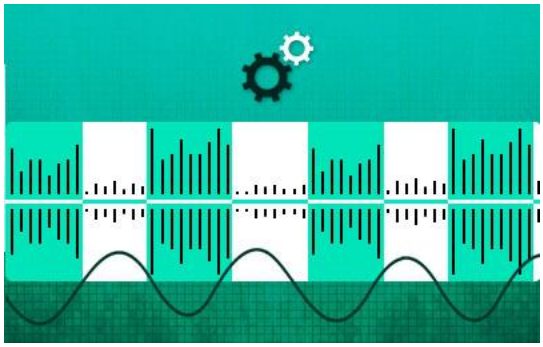
Zoom to selected clips

Quickly zoom to the time range of one or more selected clips.



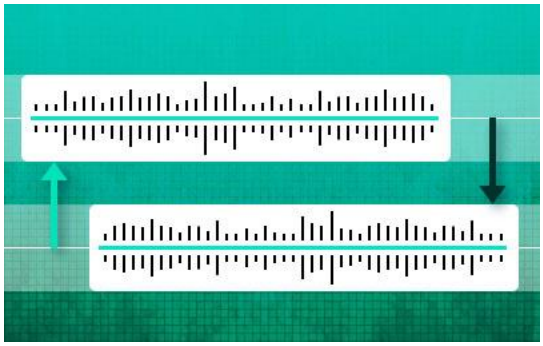
Manual keyframe entry

Edit time and parameter values for any keyframe



Auto-ducking for ambience

Now - in addition to music - choose to automatically duck ambient sound behind dialogue and SFX clips you want to stand out.



Nudge clips up and down

Use this feature to move selected clips up and down a track.

See detailed new feature summary

Previous releases of Audition

- Feature Summary | Audition | 2019 releases
- Feature Summary | Audition | 2018 releases

Audition system requirements

Applicable for November 2019 (13.0) release

System requirements for earlier releases Audition system requirements | earlier releases

Minimum system requirements for Audition

Windows

	Minimum requirement
Processor	Multicore processor with 64-bit support
Operating system	Microsoft® Windows 10 (64 bit) version 1703 or higher.
RAM	4GB of RAM
Hard disk space	4GB of available hard-disk space for installation; plus additional free space required during installation (cannot install on removable flash storage devices)
Monitor resolution	1920x1080 or larger display
OpenGL	OpenGL 2.0 capable system
Sound card	Sound card compatible with ASIO protocol, WASAPI, or Microsoft WDM/MME
Control surface support	USB interface and/or MIDI interface may be required for external control surface support (see manufacturer's specifications for your device)
CD burning	Optical drive is used for CD burning (optional)
Internet	Internet connection and registration are necessary for required software activation, validation of subscriptions, and access to online services.

macOS

	Minimum requirement
Processor	Multicore processor with 64-bit support
Operating system	macOS X Version 10.12 or higher
RAM	4GB of RAM
Hard disk space	4GB of available hard-disk space for installation; additional free space required during installation (cannot install on a volume that uses a case-sensitive file system or on removable flash storage devices)
Display	1920x1080 or larger display (for best results)
OpenGL	OpenGL 2.0–capable system; optional GPU for optimum video playback performance
Control surface support	USB interface and/or MIDI interface may be required for external control surface support (see manufacturer's specifications for your device)
CD burning	Optical drive is used for CD burning (optional)
Internet	Internet connection and registration are necessary for required software activation, validation of subscriptions, and access to online services.

*This product may integrate with or allow access to certain Adobe or third-party hosted online services. Adobe online services, including the Adobe Creative Cloud service, are available only to users 13 and older and require agreement to additional terms and Adobe's online privacy policy. The applications and online services are not available in all countries or languages, may require user registration, and may be subject to change or discontinuation without notice. Additional fees or membership charges may apply.

Language versions available for Audition

French

German

Japanese

Italian

Spanish

Portuguese

Simplified Chinese

Korean

Customizing workspaces

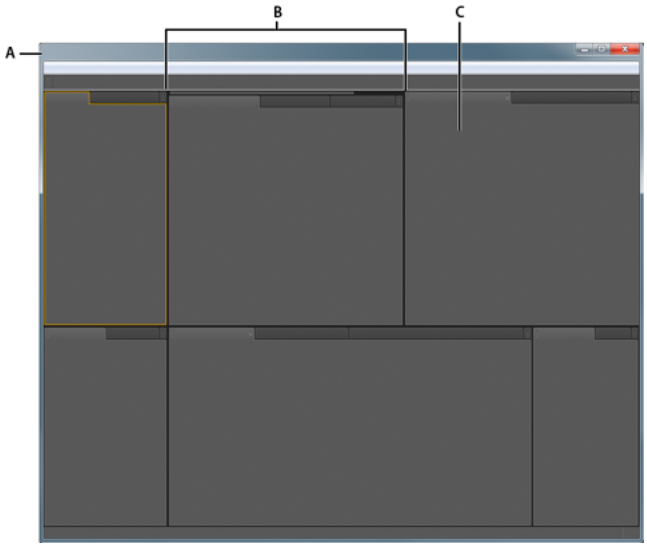
About workspaces

Adobe video and audio applications provide a consistent, customizable workspace. Although each application has its own set of panels (such as Project, Metadata, and Timeline), you move and group panels in the same way across products.

The main window of a program is the *application window*. Panels are organized in this window in an arrangement called a *workspace*. The default workspace contains groups of panels as well as panels that stand alone.

You customize a workspace by arranging panels in the layout that best suits your working style. As you rearrange panels, the other panels resize automatically to fit the window. You can create and save several custom workspaces for different tasks—for example, one for editing and one for previewing.

You can use floating windows to create a workspace more like workspaces in previous versions of Adobe applications, or to place panels on multiple monitors.



Example workspace
***A** Application window **B** Grouped panels **C** Individual panel*

Choose a workspace

Each Adobe video and audio application includes several predefined workspaces that optimize the layout of panels for specific tasks. When you choose one of these workspaces, or any custom workspaces you've saved, the current workspace is redrawn accordingly.

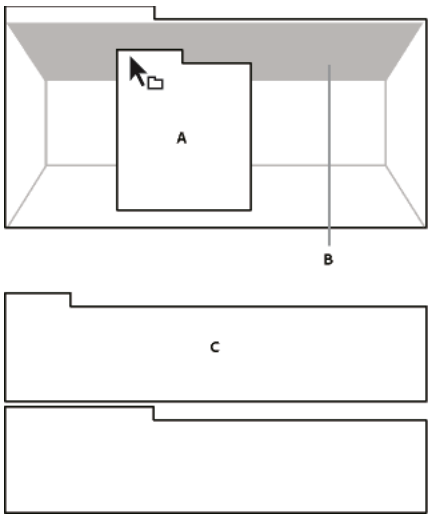
? Open the project you want to work on, choose Window > Workspace, and select the desired workspace.

Dock, group, or float panels

You can dock panels together, move them into or out of groups, and undock them so they float above the application window. As you drag a panel, *drop zones*—areas onto which you can move the panel—become highlighted. The drop zone you choose determines where the panel is inserted, and whether it docks or groups with other panels.

Docking zones

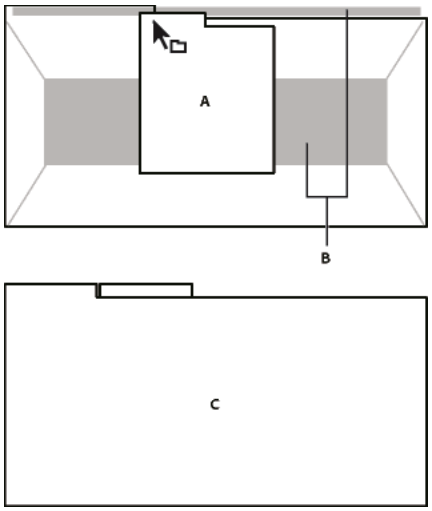
Docking zones exist along the edges of a panel, group, or window. Docking a panel places it adjacent to the existing group, resizing all groups to accommodate the new panel.



Dragging panel (A) onto docking zone (B) to dock it (C)

Grouping zones

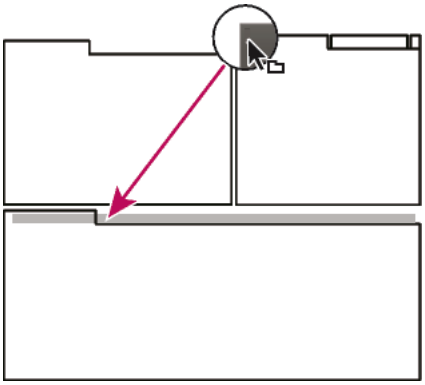
Grouping zones exist in the middle of a panel or group, and along the tab area of panels. Dropping a panel on a grouping zone stacks it with other panels.



Dragging panel (A) onto grouping zone (B) to group it with existing panels (C)

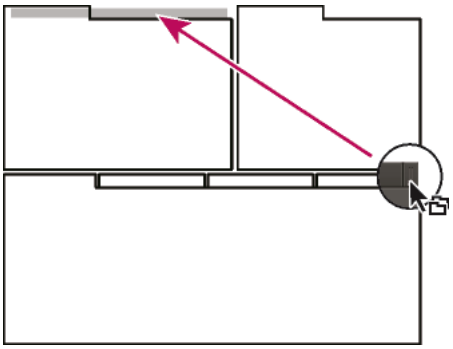
Dock or group panels

1. If the panel you want to dock or group is not visible, choose it from the Window menu.
2. Do one of the following:
   - To move an individual panel, drag the gripper area in the upper-left corner of a panel's tab onto the desired drop zone.



Drag panel gripper to move one panel

- To move an entire group, drag the group gripper in the upper-right corner onto the desired drop zone.



Drag group gripper to move entire group

The application docks or groups the panel, according to the type of drop zone.

Undock a panel in a floating window

When you undock a panel in a floating window, you can add panels to the window and modify it similarly to the application window. You can use floating windows to use a secondary monitor, or to create workspaces like the workspaces in earlier versions of Adobe applications.

? Select the panel you want to undock (if it's not visible, choose it from the Window menu), and then do one of the following:

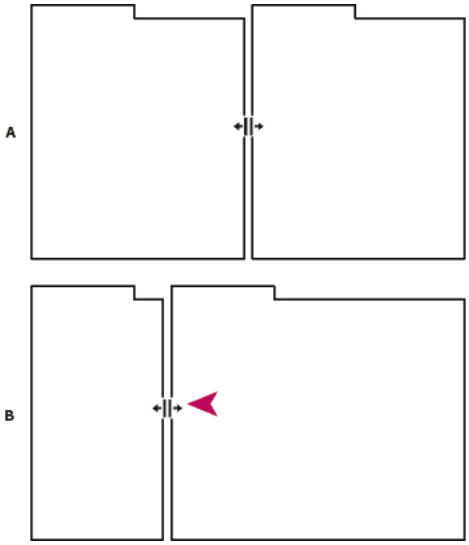
- Choose Undock Panel or Undock Frame from the panel menu. Undock Frame undocks the panel group.
- Hold down Ctrl (Windows®) or Command (Mac OS®), and drag the panel or group from its current location. When you release the mouse button, the panel or group appears in a new floating window.
- Drag the panel or group outside the application window. (If the application window is maximized, drag the panel to the Windows taskbar.)

Resize panel groups

When you position the pointer over dividers between panel groups, resize icons appear. When you drag these icons, all groups that share the divider are resized. For example, suppose your workspace contains three panel groups stacked vertically. If you drag the divider between the bottom two groups, they are resized, but the topmost group doesn't change.

To quickly maximize a panel beneath the pointer, press the accent key. (Do not press Shift.) Press the accent key again to return the panel to its original size.

1. Do either of the following:
   - To resize either horizontally or vertically, position the pointer between two panel groups. The pointer becomes a double-arrow .
   - To resize in both directions at once, position the pointer at the intersection between three or more panel groups. The pointer becomes a four-way arrow .
2. Hold down the mouse button, and drag to resize the panel groups.



Dragging divider between panel groups to resize them horizontally
***A** Original group with resize icon **B** Resized groups*

Open, close, and scroll to panels

When you close a panel group in the application window, the other groups resize to use the newly available space. When you close a floating window, the panels within it close, too.

- To open a panel, choose it from the Window menu.
- To close a panel or window, press Control-W (Windows) or Command-W (Mac OS), or click its Close button .
- To see all the panel tabs in a narrow panel group, drag the horizontal scroll bar.
- To bring a panel to the front of a group of panels, do one of the following:
  - Click the tab of the panel you want in front.

- Hover the cursor above the tab area, and turn the mouse scroll wheel. Scrolling brings each panel to the front, one after another.
- Drag tabs horizontally to change their order.

- To reveal panels hidden in a narrow panel group, drag the scroll bar above the panel group.

Drag horizontal scroll bar to see all panels in narrow group

Working with multiple monitors

To increase the available screen space, use multiple monitors. When you work with multiple monitors, the application window appears on one monitor, and you place floating windows on the second monitor. Monitor configurations are stored in the workspace.

Display the toolbar

The toolbar provides quick access to tools, the Workspace menu, and buttons that toggle between the Waveform and Multitrack editors. Some tools are unique to each view. Likewise, some Waveform Editor tools are available only in the spectral display.

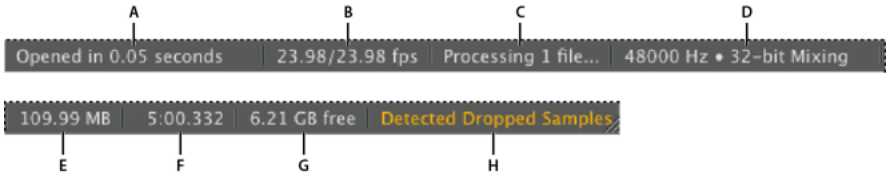
By default, the toolbar is docked immediately below the menu bar. However, you can undock the toolbar, converting it to the Tools panel, which you can manipulate like any other panel.

- To show or hide the toolbar, choose Window > Tools. A check mark by the Tools command indicates that it is shown.
- To undock the toolbar from its default location, drag the handle at the left edge to another location in the work area.
- To redock the Tools panel in its default location, drag the Tools panel tab to the drop zone that spans the entire width of the Adobe Audition window, just under the menu bar.

Available tools differ in each view.
***A** Waveform Editor tools for spectral display **B** Multitrack Editor tools*

Display the status bar

The status bar runs across the bottom of the Adobe Audition work area. The far left of the status bar indicates the time required to open, save, or process a file, as well as the current transport status (Playing, Recording, or Stopped). The far right of the bar displays various information that you can customize.



Status bar
***A** Time to open, save, or process file **B** Video Frame Rate **C** File Status **D** Sample Type **E** Uncompressed Audio Size **F** Duration **G** Free Space **H** Detect Dropped Samples*

- To show or hide the status bar, choose View > Status Bar > Show. A check mark indicates that the status bar is visible.
- To change the information displayed at the far right of the bar, choose View > Status Bar, or right-click the bar. Then select from the following options:

  Show Data Under Cusror: Displays the frequency, time, channel, and amplitude information at the location under the cursor.

  Video Frame Rate: Displays the current and target frame rate of open video files in the Multitrack Editor.

  File Status: Indicates when processing is occurring for effects and amplitude adjustments.

  Sample Type: Displays sample information about the currently opened waveform (Waveform Editor) or session file (Multitrack Editor). For example, a 44,100 Hz, 16-bit stereo file is displayed as 44100 Hz • 16-bit • Stereo.

  Uncompressed Audio Size: Indicates either how large the active audio file would be if saved to an uncompressed format such as WAV and AIFF, or the total size of a multitrack session.

  Duration: Shows you the length of the current waveform or session. For example, 0:01:247 means the waveform or session is 1.247 seconds long.

  Free Space: Shows how much space is available on your hard drive.

  Free Space (Time): Displays the time remaining for recording, based upon the currently selected sample rate. This value is shown as minutes, seconds, and thousandths of seconds. For example, if Adobe Audition is set to record 8-bit mono audio at 11,025 Hz, the time remaining might read 4399:15.527 free. Change the recording options to 16-bit stereo at 44,100 Hz, and the time remaining becomes 680:44.736 free.

  ***Tip**: By default, Free Space (Time) information is hidden. To show it, right-click the status bar, and select Free Space (Time) from the pop-up menu.*

  Detect Dropped Samples: Indicates that samples were missing during recording or playback. If this indicator appears, consider rerecording the file to avoid audible dropouts.

Change interface colors, brightness, and performance

1. Choose Edit > Preferences > Appearance (Windows) or Audition > Preferences > Appearance (Mac OS).
2. Adjust any of the following options, and then click OK:**Presets**Applies, saves, or deletes a combination of Colors and Brightness settings.**Colors**Click a swatch to change the color of waveforms, selections, or the current-time indicator.**Brightness**Brightens or darkens panels, windows, and dialog boxes.**Use Gradients**When deselected, removes shadows and highlights from panels, buttons, and meters.

Save, reset, or delete workspaces

Save a custom workspace

As you customize a workspace, the application tracks your changes, storing the most recent layout. To store a specific layout more permanently, save a custom workspace. Saved custom workspaces appear in the Workspace menu, where you can return to and reset them.

? Arrange the frames and panels as desired, and then choose Window > Workspace > New Workspace. Type a name for the workspace, and click OK.

Note:

*(After Effects, **Premiere Pro**, Encore) If a project saved with a custom workspace is opened on another system, the application looks for a workspace with a matching name. If it can't find a match (or the monitor configuration doesn't match), it uses the current local workspace.*

Reset a workspace

Reset the current workspace to return to its original, saved layout of panels.

? Choose Window > Workspace > Reset *workspace name*.

Delete a workspace

1 Choose Window > Workspace > **Delete Workspace**.

2 Choose the workspace you want to delete, and then click OK.

Note:

You cannot delete the currently active workspace.

Reduce noise and restore audio

How to Reduce Noise and Restore Audio in Adobe Audition

How to Reduce Noise and Restore Audio in Adobe Audition

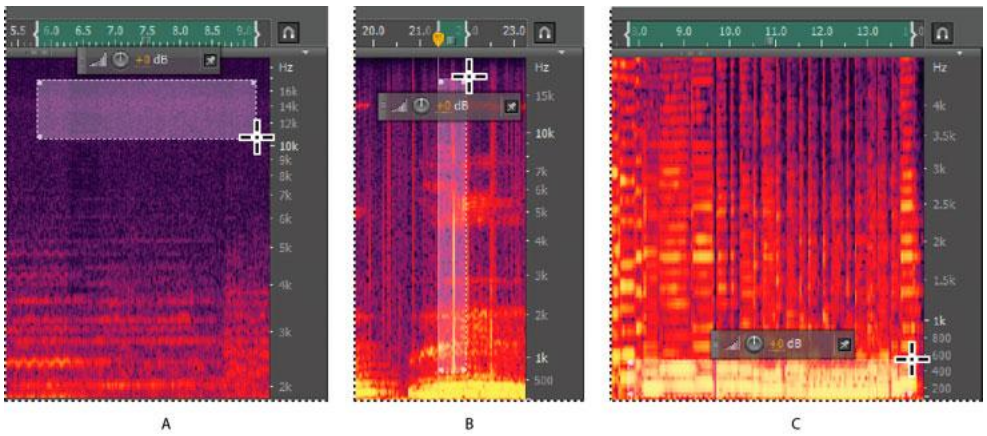
Techniques for restoring audio

You can fix a wide array of audio problems by combining two powerful features. First, use **Spectral Display** to visually identify and select ranges of noise or individual artifacts. (See Select spectral ranges and Select artifacts and repair them automatically.) Then, use either **Diagnostic** or **Noise Reduction** effects to fix problems like the following:

- Crackle from wireless microphones or old vinyl records. (See Automatic Click Remover effect.)
- Background noise like wind rumble, tape hiss, or power-line hum. (See Adaptive Noise Reduction effectand DeHummer effect.)
- Phase cancelation from poorly placed stereo microphones or misaligned tape machines. (See Automatic Phase Correction effect.)

*The real-time restoration effects above, which are available in both the **Waveform** and **Multitrack** editors, quickly address common audio problems. For unusually noisy audio, however, consider using offline, process effects unique to the **Waveform Editor**, such as **Hiss Reduction** and **Noise Reduction**.*

Watch the Audio restoration techniques video to learn best practices for fixing audio in Audition using the Amplitude Statistics panel, spectral frequency display, adaptive noise reduction, Diagnostics panel, and DeClipper and DeHummer effects.



Selecting various types of noise in Spectral Display
***A** Hiss **B** Crackle **C** Rumble*

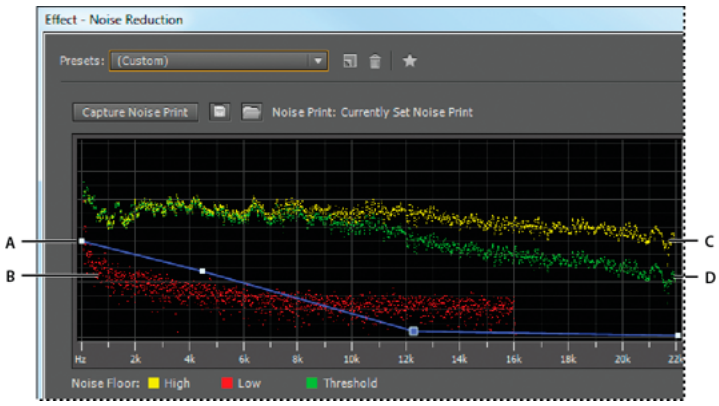
Watch the video How to use the Spectral Frequency Display to clean up your audio to learn more about using Spectral Frequency Display.

Noise Reduction effect (Waveform Editor only)

The **Noise Reduction/Restoration** > **Noise Reduction** effect dramatically reduces background and broadband noise with a minimal reduction in signal quality. This effect can remove a combination of noise, including tape hiss, microphone background noise, power-line hum, or any noise that is constant throughout a waveform.

The proper amount of noise reduction depends upon the type of background noise and the acceptable loss in quality for the remaining signal. In general, you can increase the signal-to-noise ratio by 5 to 20 dB and retain high audio quality.

To achieve the best results with the **Noise Reduction** effect, apply it to audio with no DC offset. With a DC offset, this effect may introduce clicks in quiet passages. (To remove a DC offset, choose **Favorites** > **Repair DC Offset**.)



Evaluating and adjusting noise with the Noise Reduction graph:
***A** Drag control points to vary reduction in different frequency ranges **B** Low amplitude noise. **C** High amplitude noise **D** Threshold below which noise reduction occurs.*

Apply the Noise Reduction effect

1 In the **Waveform Editor**, select a range that contains only noise and is at least half a second long.

*To select noise in a specific frequency range, use the **Marquee Selection** tool. (See Select spectral ranges.)*

2 Choose **Effects** > **Noise Reduction/Restoration** > **Capture Noise Print**.

3 In the **Editor** panel, select the range from which you want to remove noise.

4 Choose **Effects** > **Noise Reduction/Restoration** > **Noise Reduction**.

5 Set the desired options.

When recording in noisy environments, record a few seconds of representative background noise that can be used as a noise print later on.

Noise Reduction options

Capture Noise Print Extracts a noise profile from a selected range, indicating only background noise. Adobe Audition gathers statistical information about the background noise so it can remove it from the remainder of the waveform.

Tip: If the selected range is too short, **Capture Noise Print** is disabled. Reduce the **FFT Size** or select a longer range of noise. If you can't find a longer range, copy and paste the currently selected range to create one. (You can later remove the pasted noise by using the **Edit** > **Delete** command.)

Save the Current Noise Print

Saves the noise print as an .fft file, which contains information about sample type, FFT (Fast Fourier Transform) size, and three sets of FFT coefficients: one for the lowest amount of noise found, one for the highest amount, and one for the power average.

Load a Noise Print from Disk

Opens any noise print previously saved from Adobe Audition in FFT format. However, you can apply noise prints only to identical sample types. (For example, you can't apply a 22 kHz mono profile to 44kHz stereo samples.)

note: Because noise prints are so specific, a print for one type of noise won't produce good results with other types. If you regularly remove similar noise, however, a saved profile can greatly increase efficiency.

Graph Depicts frequency along the x-axis (horizontal) and the amount of noise reduction along the y-axis (vertical).

The blue control curve sets the amount of noise reduction in different frequency ranges. For example, if you need noise reduction only in the higher frequencies, adjust the control curve downward to the right of the graph.

If you click the Reset button to flatten the control curve, the amount of noise reduction is based entirely on the noise print.

Tip: To better focus on the noise floor, click the menu button to the upper right of the graph, and deselect **Show Control Curve** and **Show Tooltip Over Graph**.

Noise Floor High shows the highest amplitude of detected noise at each frequency; Low shows the lowest amplitude. Threshold shows the amplitude below which noise reduction occurs.

Tip: The three elements of the noise floor can overlap in the graph. To better distinguish them, click the menu button , and select options from the **Show Noise Floor** menu.

Scale Determines how frequencies are arranged along the horizontal x-axis:

- For finer control over low frequencies, select **Logarithmic**. A logarithmic scale more closely resembles how people hear sound.
- For detailed, high-frequency work with evenly spaced intervals in frequency, select **Linear**.

Channel Displays the selected channel in the graph. The amount of noise reduction is always the same for all channels.

Select Entire File Lets you apply a captured noise print to the entire file.

Noise Reduction Controls the percentage of noise reduction in the output signal. Fine-tune this setting while previewing audio to achieve maximum noise reduction with minimum artifacts. (Excessively high noise reduction levels can sometimes cause audio to sound flanged or out-of-phase.)

Reduce By Determines the amplitude reduction of detected noise. Values between 6 and 30 dB work well. To reduce bubbly artifacts, enter lower values.

Output Noise Only Previews only noise so you determine if the effect is removing any desirable audio.

Advanced settings Click the triangle to display the following options:

Spectral Decay Rate Specifies the percentage of frequencies processed when audio falls below the noise floor. Fine-tuning this percentage allows greater noise reduction with fewer artifacts. Values of 40% to 75% work best. Below those values, bubbly-sounding artifacts are often heard; above those values, excessive noise typically remains.

Smoothing Takes into account the variance of the noise signal in each frequency band. Bands that vary greatly when analyzed (such as white noise) will be smoothed differently than constant bands (like 60-Hz hum). In general, increasing the smoothing amount (up to 2 or so) reduces burbly background artifacts at the expense of raising the overall background broadband noise level.

Precision Factor Controls changes in amplitude. Values of 5-10 work best, and odd numbers are ideal for symmetrical processing. With values of 3 or less, the Fast Fourier transform is performed in giant blocks, and between them drops or spikes in volume can occur. Values beyond 10 cause no noticeable change in quality, but they increase processing time.

Transition Width Determines the amplitude range between noise and desirable audio. For example, a width of zero applies a sharp, noise gate to each frequency band. Audio just above the threshold remains; audio just below is truncated to silence. Alternatively, you can specify a range over which the audio fades to silence based upon the input level. For

example, if the transition width is 10 dB, and the noise level for the band is -60 dB, audio at -60 dB stays the same, audio at -62 dB is reduced slightly, and audio at -70 dB is removed entirely.

FFT Size Determines how many individual frequency bands are analyzed. This option causes the most drastic changes in quality. The noise in each frequency band is treated separately, so with more bands, noise is removed with finer frequency detail. Good settings range from 4096 to 8192.

Fast Fourier Transform size determines the tradeoff between frequency- and time-accuracy. Higher FFT sizes might cause swooshing or reverberant artifacts, but they very accurately remove noise frequencies. Lower FFT sizes result in better time response (less swooshing before cymbal hits, for example), but they can produce poorer frequency resolution, creating hollow or flanged sounds.

Noise Print Snapshots Determines how many snapshots of noise to include in the captured profile. A value of 4000 is optimal for producing accurate data.

Very small values greatly affect the quality of the various noise reduction levels. With more snapshots, a noise reduction level of 100 will likely cut out more noise, but also cut out more original signal. However, a low noise reduction level with more snapshots will also cut out more noise, but likely retain the intended signal.

Sound Remover effect

The **Sound Remover** effect (**Effects > Noise Reduction/Restoration**) removes unwanted audio sources from a recording. This effect analyzes a selected portion of the recording, and builds a sound model, which is used to find and remove the sound.

The generated model can also be modified using parameters that indicate its complexity. A high complexity sound model requires more refinement passes to process the recording, but provides more accurate results. You can also save the sound model for later use. Several common presets are also included to remove some common noise sounds, such as sirens and ringing mobile phones.

Learn Sound Model Uses the selected waveform to learn the sound model. Select an area on the waveform that only contains the sound to remove, and then press Learn Sound Model. You can also save and load sound models on disc.

Sound Model Complexity Indicates the complexity of the **Sound Model**. The more complex or mixed the sound is, the better results you'll get with a higher complexity setting, though the longer it will take to calculate. Settings range from 1 to 100.

Sound Refinement Passes Defines the number of refinement passes to make to remove the sound patterns indicated in the sound model. Higher number of passes require longer processing time, but offer more accurate results.

Content Complexity Indicates the complexity of the signal. The more complex or mixed the sound is, the better results you'll get with a higher complexity setting, though the longer it will take to calculate. Settings range from 5 to 100.

Content Refinement Passes Specifies the number of passes to make on the content to remove the sounds that match the sound model. A higher number of passes require more processing time, but generally provide more accurate results.

Enhanced Supression This increases the aggressiveness of the sound removal algorithm, and can be modified on the **Strength** value. A higher value will remove more of the sound model from mixed signals, which can result in greater loss of desired signal, while a lower value will leave more of the overlapping signal and therefore, more of the noise may be audible (though less than the original recording.)

Enhance for Speech Specifies that the audio includes speech and is careful in removing audio patterns that closely resemble speech. The end result makes sure that speech is not removed, while removing noise.

FFT Size Determines how many individual frequency bands are analyzed. This option causes the most drastic changes in quality. The noise in each frequency band is treated separately, so with more bands, noise is removed with finer frequency detail. Good settings range from 4096 to 8192. Fast Fourier Transform size determines the tradeoff between

frequency- and time-accuracy. Higher FFT sizes might cause swooshing or reverberant artifacts, but they very accurately remove noise frequencies. Lower FFT sizes result in better time response (less swooshing before cymbal hits, for example), but they can produce poorer frequency resolution, creating hollow or flanged sounds.

Watch the video Sound removal and noise reduction strategies to see how you can reduce noise and remove unwanted sounds from your audio.

Adaptive Noise Reduction effect

The **Noise Reduction/Restoration** > **Adaptive Noise Reduction** effect quickly removes variable broadband noise such as background sounds, rumble, and wind. Because this effect operates in real time, you can combine it with other effects in the **Effects Rack** and apply it in the **Multitrack Editor**. By contrast, the standard **Noise Reduction** effect is available only as an offline process in the **Waveform Editor**. That effect, however, is sometimes more effective at removing constant noise, such as hiss or hum.

For best results, apply **Adaptive Noise Reduction** to selections that begin with noise followed by desirable audio. The effect identifies noise based on the first few seconds of audio.

Note:

*This effect requires significant processing. If your system performs slowly, lower **FFT Size** and turn off **High Quality Mode**.*

Reduce Noise By Determines the level of noise reduction. Values between 6 and 30 dB work well. To reduce bubbly background effects, enter lower values.

Noisiness Indicates the percentage of original audio that contains noise.

Fine Tune Noise Floor Manually adjusts the noise floor above or below the automatically calculated floor.

Signal Threshold Manually adjusts the threshold of desirable audio above or below the automatically calculated threshold.

Spectral Decay Rate Determines how quickly noise processing drops by 60 decibels. Fine-tuning this setting allows greater noise reduction with fewer artifacts. Values that are too short create bubbly sounds; values that are too long create a reverb effect.

Broadband Preservation Retains desirable audio in specified frequency bands between found artifacts. A setting of 100 Hz, for example, ensures that no audio is removed 100 Hz above or below found artifacts. Lower settings remove more noise but may introduce audible processing.

FFT Size Determines how many individual frequency bands are analyzed. Choose a high setting to increase frequency resolution; choose a low setting to increase time resolution. High settings work well for artifacts of long duration (like squeaks or power-line hum), while low settings better address transient artifacts (like clicks and pops).

Watch the video Sound removal and noise reduction strategies to see how you can reduce noise and remove unwanted sounds from your audio.

Automatic Click Remover effect

To quickly remove crackle and static from vinyl recordings, use the **Noise Reduction/Restoration** > **Automatic Click Remover** effect. You can correct a large area of audio or a single click or pop.

This effect provides the same options as the **DeClicker** effect, which lets you choose which detected clicks to address (see DeClicker options). However, because the **Automatic Click Remover** operates in real time, you can combine it with other effects in the **Effects Rack** and apply it in the **Multitrack Editor**. The **Automatic Click Remover** effect also applies multiple scan and repair passes automatically; to achieve the same level of click reduction with the **DeClicker**, you must manually apply it multiple times.

Threshold Determines sensitivity to noise. Lower settings detect more clicks and pops but may include audio you wish to retain. Settings range from 1 to 100; the default is 30.

Complexity Indicates the complexity of noise. Higher settings apply more processing but can degrade audio quality. Settings range from 1 to 100; the default is 16.

Automatic Phase Correction effect

The **Noise Reduction/Restoration** > **Automatic Phase Correction** effect addresses azimuth errors from misaligned tape heads, stereo smearing from incorrect microphone placement, and many other phase-related problems.

Global Time Shift Activates the **Left** and **Right Channel Shift** sliders, which let you apply a uniform phase shift to all selected audio.

Auto Align Channels and Auto Center Panning Align phase and panning for a series of discrete time intervals, which you specify using the following options:

Time Resolution Specifies the number of milliseconds in each processed interval. Smaller values increase accuracy; larger ones increase performance.

Responsiveness Determines overall processing speed. Slow settings increase accuracy; fast settings increase performance.

Channel Specifies the channels phase correction will be applied to.

Analysis Size Specifies the number of samples in each analyzed unit of audio.

*For the most precise, effective phase correction, use the **Auto Align Channels** option. Enable the **Global Time Shift** sliders only if you are confident that a uniform adjustment is necessary, or if you want to manually animate phase correction in the **Multitrack Editor**.*

Click/Pop Eliminator effect

Use the **Click/Pop Eliminator** effect (**Effects > Noise Reduction/Restoration**) to remove microphone pops, clicks, light hiss, and crackles. Such noise is common on recordings such as old vinyl records and on-location recordings. The effect dialog box stays open, and you can adjust the selection, and fix multiple clicks without reopening the effect several times.

Detection and correction settings are used to find clicks and pops. The detection and rejection ranges are displayed graphically.

Detection graph Shows the exact threshold levels to be used at each amplitude, with amplitude along the horizontal ruler (x-axis) and threshold level along the vertical ruler (y-axis). Adobe Audition uses values on the curve to the right (above -20 dB or so) when processing louder audio and values on the left when processing softer audio. Curves are color-coded to indicate detection and rejection.

Scan for All Levels Scans the highlighted area for clicks based on the values for **Sensitivity** and **Discrimination**, and determines values for **Threshold**, **Detect**, and **Reject**. Five areas of audio are selected, starting at the quietest and moving to the loudest.

Sensitivity Determines the level of clicks to detect. Use a lower value, such as 10, to detect lots of subtle clicks, or a value of 20 to detect a few louder clicks. (Detected levels with **Scan for All Levels** are always higher than with this option.)

Discrimination Determines how many clicks to fix. Enter high values to fix very few clicks and leave most of the original audio intact. Enter lower values, such as 20 or 40, if the audio contains a moderate number of clicks. Enter extremely low values, such as 2 or 4, to fix constant clicks.

Scan for Threshold Levels Automatically sets the Maximum, Average, and Minimum Threshold levels.

Maximum, Average, Minimum Determine the unique detection and rejection thresholds for the maximum, average, and minimum amplitudes of the audio. For example, if audio has a maximum RMS amplitude of -10 dB, you should set **Maximum Threshold** to -10 dB. If the minimum RMS amplitude is -55 dB, then set **Minimum Threshold** to -55.

Set the threshold levels before you adjust the corresponding **Detect** and **Reject** values. (Set the **Maximum** and **Minimum Threshold** levels first, because once they're in place, you shouldn't need to adjust them much.) Set the **Average Threshold** level to about three quarters of the way between the **Maximum** and **Minimum Threshold** levels. For example, if **Maximum Threshold** is set to 30 and **Minimum Threshold** is set to 10, set **Average Threshold** to 25.

After you audition a small piece of repaired audio, you can adjust the settings as needed. For example, if a quiet part still has a lot of clicks, lower the **Minimum Threshold** level a bit. If a loud piece still has clicks, lower the **Average** or **Maximum Threshold** level. In general, less correction is required for louder audio, as the audio itself masks many clicks, so repairing them isn't necessary. Clicks are very noticeable in very quiet audio, so quiet audio tends to require lower detection and rejection thresholds.

Second Level Verification (Reject Clicks) Rejects some of the potential clicks found by the click detection algorithm. In some types of audio, such as trumpets, saxophones, female vocals, and snare drum hits, normal peaks are sometimes detected as clicks. If these peaks are corrected, the resulting audio will sound muffled. **Second Level Verification** rejects these audio peaks and corrects only true clicks.

Detect Determines sensitivity to clicks and pops. Possible values range from 1 to 150, but recommended values range from 6 to 60. Lower values detect more clicks.

Start with a threshold of 35 for high-amplitude audio (above -15 dB), 25 for average amplitudes, and 10 for low-amplitude audio (below-50 dB). These settings allow for the most clicks to be found, and usually all of the louder ones. If a constant crackle is in the background of the source audio, try lowering the **Min Threshold** level or increasing the dB level to which the threshold is assigned. The level can be as low as 6, but a lower setting can cause the filter to remove sound other than clicks.

If more clicks are detected, more repair occurs, increasing the possibility of distortion. With too much distortion of this type, audio begins to sound flat and lifeless. If this occurs, set the detection threshold rather low, and select **Second Level Verification** to reanalyze the detected clicks and disregard percussive transients that aren't clicks.

Reject Determines how many potential clicks (found using the **Detection Threshold**) are rejected if **Second Level Verification** box is selected. Values range from 1 to 100; a setting of 30 is a good starting point. Lower settings allow for more clicks to be repaired. Higher settings can prevent clicks from being repaired, as they might not be actual clicks.

You want to reject as many detected clicks as possible but still remove all audible clicks. If a trumpet-like sound has clicks in it, and the clicks aren't removed, try lowering the value to reject fewer potential clicks. If a particular sound becomes distorted, then increase the setting to keep repairs at a minimum. (The fewer repairs that are needed to get good results, the better.)

FFT Size Determines the FFT size used to repair clicks, pops, and crackle. In general, select **Auto** to let Adobe Audition determine the FFT size. For some types of audio, however, you might want to enter a specific FFT size (from 8 to 512).

A good starting value is 32, but if clicks are still quite audible, increase the value to 48, and then 64, and so on. The higher the value, the slower the correction will be, but the better the potential results. If the value is too high, rumbly, low frequency distortion can occur.

Fill Single Click Corrects a single click in a selected audio range. If Auto is selected next to **FFT Size**, then an appropriate FFT size is used for the restoration based on the size of the area being restored. Otherwise, settings of 128 to 256 work very well for filling in single clicks. Once a single click is filled, press the F3 key to repeat the action. You can also create a quick key in the **Favorites** menu for filling in single clicks.

Pop Oversamples Width Includes surrounding samples in detected clicks. When a potential click is found, its beginning and end points are marked as closely as possible. The **Pop Oversamples** value (which can range from 0 to 300) expands that range, so more samples to the left and right of the click are considered part of the click. If corrected clicks become quieter but are still evident, increase the **Pop oversamples** value. Start with a value of 8, and increase it slowly to as much as 30 or 40. Audio that doesn't contain a click shouldn't change very much if it's corrected, so this buffer area should remain mostly untouched by the replacement algorithm. Increasing the **Pop Oversamples** value also forces larger FFT sizes to be used if **Auto** is selected. A larger setting may remove clicks more cleanly, but if it's too high, audio will start to distort where the clicks are removed.

Run Size Specifies the number of samples between separate clicks. Possible values range from 0 to 1000. To independently correct extremely close clicks, enter a low value; clicks that occur within the **Run Size** range are corrected together.

A good starting point is around 25 (or half the FFT size if **Auto** next to **FFT Size** isn't selected). If the **Run Size** value is too large (over 100 or so), then the corrections may become more noticeable, as very large blocks of data are repaired at once. If you set the **Run Size** too small, then clicks that are very close together may not be repaired completely on the first pass.

Pulse Train Verification Prevents normal waveform peaks from being detected as clicks. It may also reduce detection of valid clicks, requiring more aggressive threshold settings. Select this option only if you've already tried to clean up the audio but stubborn clicks remain.

Link Channels Processes all channels equally, preserving the stereo or surround balance. For example, if a click is found in one channel, a click will most likely be detected in the other.

Detect Big Pops Removes large unwanted events (such as those more than a few hundred samples wide) that might not be detected as clicks. Values can range from 30 to 200.

Note that a sharp sound like a loud snare drum hit can have the same characteristic as a very large pop, so select this option only if you know the audio has very large pops (like a vinyl record with a very big scratch in it). If this option causes drum hits to sound softer, slightly increase the threshold to fix only loud, obvious pops.

If loud, obvious pops aren't fixed, select **Detect Big Pops**, and use settings from about 30 (to find quiet pops) to 70 (to find loud pops).

Ignore Light Crackle Smooths out one-sample errors when detected, often removing more background crackle. If the resulting audio sounds thinner, flatter, or more tinny, deselect this option.

Passes Performs up to 32 passes automatically to catch clicks that might be too close together to be repaired effectively. Fewer passes occur if no more clicks are found and all detected clicks are repaired. In general, about half as many clicks are repaired on each successive pass. A higher detection threshold might lead to fewer repairs and increase the quality while still removing all clicks.

Watch the video Use the Click/Pop Eliminator and DeClicker effects to learn how you can remove microphone pops, clicks, light hiss, and crackles.

DeHummer effect

The **Noise Reduction/Restoration** > **DeHummer** effect removes narrow frequency bands and their harmonics. The most common application addresses power line hum from lighting and electronics. But the **DeHummer** can also apply a notch filter that removes an overly resonant frequency from source audio.

*To quickly address typical audio problems, choose an option from the **Presets** menu.*

Frequency Sets the root frequency of the hum. If you're unsure of the precise frequency, drag this setting back and forth while previewing audio.

Note:

To visually adjust root frequency and gain, drag directly in the graph.

Q Sets the width of the root frequency and harmonics above. Higher values affect a narrower range of frequencies, and lower values affect a wider range.

Gain Determines the amount of hum attenuation.

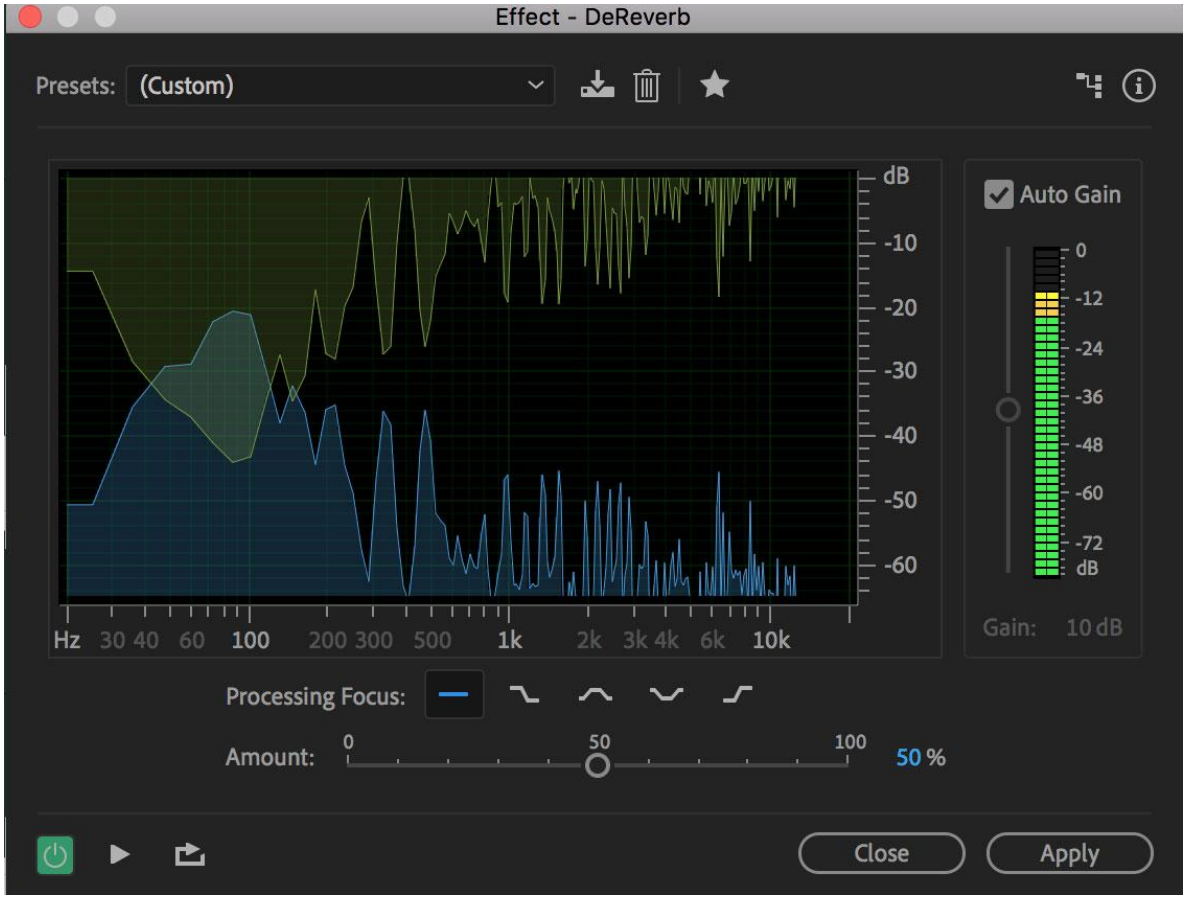
Number of Harmonics Specifies how many harmonic frequencies to affect.

Harmonic Slope Changes the attenuation ratio for harmonic frequencies.

Output Hum Only Lets you preview removed hum to determine if it contains any desirable audio.

DeReverb effect

The **Noise Reduction/Restoration > DeReverb** effect estimates the reverberation profile and helps adjust the reverberation amount. The values range from 0% to 100% and control the amount of processing applied to the audio signal.



DeReverb effect controls

Processing focus

There are five processing focus buttons. Each of the Processing focus buttons focuses the noise suppression process on specific parts of the s frequency spectrum.

All frequency focus Use this to apply the same processing to the full frequency spectrum of the signal

Hi frequency focus Use this to focus processing on the high-end range of the frequency spectrum

Hi/low frequency focus Use this focus to process more on the high and low-end range of the frequency spectrum of the signa s on the mid range

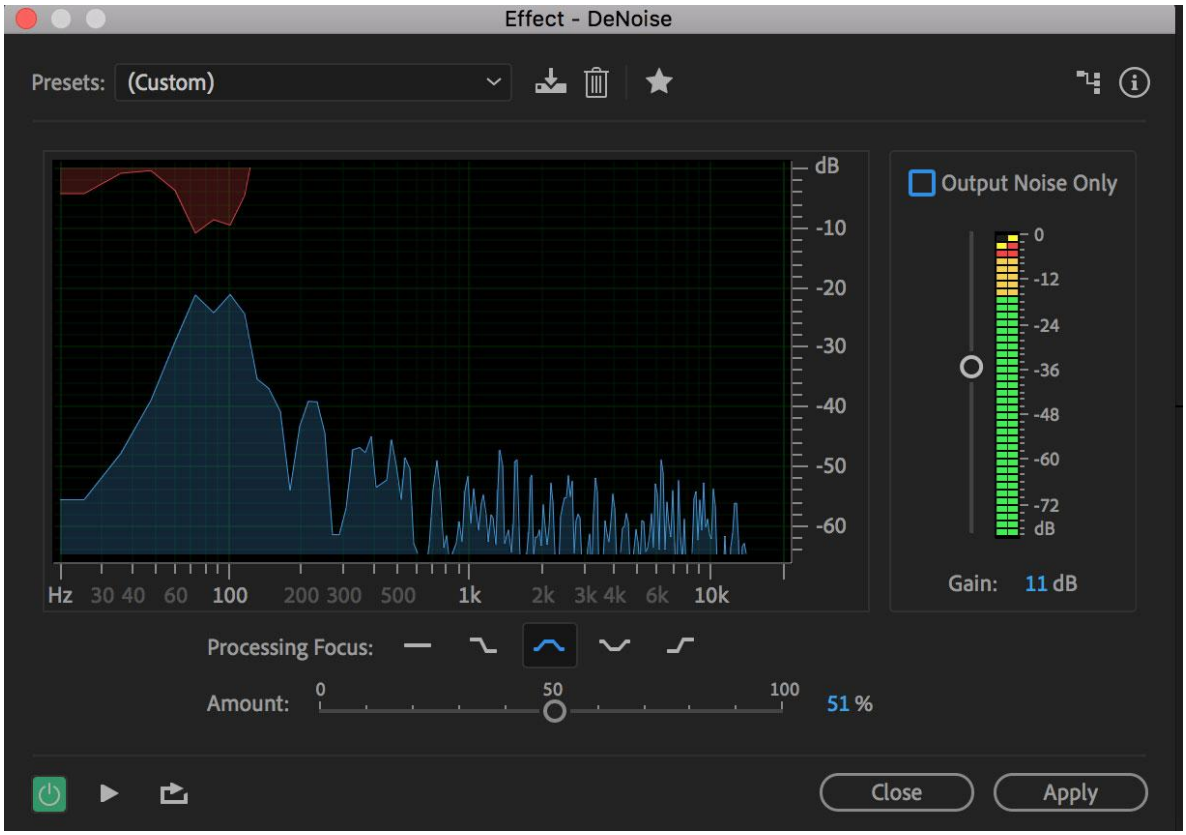
Mid frequency focus Use this option to apply focus on the mid-range of the frequency spectrum of the signal and less to the high ar nd range

Low frequency focus This option focuses processing on the low-end range of the frequency spectrum

Applying the dereverberation effect could result in lower levels of output in comparison to the orginal audio due to reduction in dynamic range. The output gain works as a make-up gain and allows you to adjust the level of the output signal. Use the slidrer to adjust gain manually. Alternatively, you can enable automatic adjustment of gain by enablig the **Auto Gain** checkbox.

DeNoise effect

The **Noise Reduction/Restoration** > **DeNoise** effect reduces or completely removes noise from your audio file. This could be unwanted hum and hiss, fans, air conditioner or any other background noise. You can control the amount of noise reduced using a slider. The values range from 0% to 100% and control the amount of processing applied to the audio signal.



DeNoise effect controls

Adjust Gain

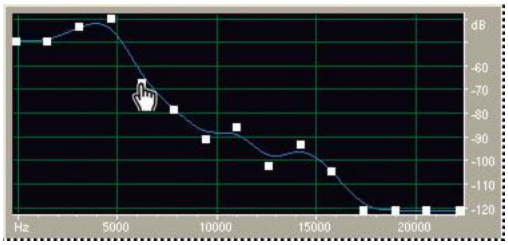
Applying the DeNoise effect could reduce the level of the output signal and make it lower than the level of original audio. Use the Gain slider to control the amount of output signal. Enable the **Output Noise Only** checkbox to listen to the removed noise in isolation.

The processing focus for DeNoise effect is similar to DeReverb effect. For more information, see Processing focus.

Hiss Reduction effect (Waveform Editor only)

The **Noise Reduction/Restoration** > **Hiss Reduction** effect reduces hiss from sources such as audio cassettes, vinyl records, or microphone preamps. This effect greatly lowers the amplitude of a frequency range if it falls below an amplitude threshold called the noise floor. Audio in frequency ranges that are louder than the threshold remain untouched. If audio has a consistent level of background hiss, that hiss can be removed completely.

To reduce other types of noise that have a wide frequency range, try the ***Noise Reduction*** *effect. (See Noise Reduction effect (Waveform Editor only).)*



Using the Hiss Reduction graph to adjust the noise floor

Capture Noise Floor Graphs an estimate of the noise floor. The estimate is used by the **Hiss Reduction** effect to more effectively remove only hiss while leaving regular audio untouched. This option is the most powerful feature of **Hiss Reduction**.

To create a graph that most accurately reflects the noise floor, click **Get Noise Floor** with a selection of audio that contains only hiss. Or, select an area that has the least amount of desirable audio, in addition to the least amount of high frequency information. (In the spectral display, look for an area without any activity in the top 75% of the display.)

After you capture the noise floor, you might need to lower the control points on the left (representing the lower frequencies) to make the graph as flat as possible. If music is present at any frequency, the control points around that frequency will be higher than they should be.

Graph Represents the estimated noise floor for each frequency in the source audio, with frequency along the horizontal ruler (x-axis) and the amplitude of the noise floor along the vertical ruler (y-axis). This information helps you distinguish hiss from desirable audio data.

The actual value used to perform hiss reduction is a combination of the graph and the **Noise Floor** slider, which shifts the estimated noise floor reading up or down for fine-tuning.

Note:

To disable tooltips for frequency and amplitude, click the menu button *to the upper right of the graph, and deselect Show Tooltip Over Graph.*

Scale Determines how frequencies are arranged along the horizontal x-axis:

- For finer control over low frequencies, select **Logarithmic**. A logarithmic scale more closely resembles how people hear sound.
- For detailed, high-frequency work with evenly spaced intervals in frequency, select **Linear**.

Channel Displays the selected audio channel in the graph.

Reset

Resets the estimated noise floor. To reset the floor higher or lower, click the menu button to the upper right of the graph, and choose an option from the **Reset Control Curve** menu.

Note:

For quick, general-purpose hiss reduction, a complete noise floor graph isn't always necessary. In many cases, you can simply reset the graph to an even level and manipulate the Noise Floor slider.

Noise Floor Fine-tunes the noise floor until the appropriate level of hiss reduction and quality is achieved.

Reduce By Sets the level of hiss reduction for audio below the noise floor. With higher values (especially above 20 dB) dramatic hiss reduction can be achieved, but the remaining audio might become distorted. With lower values, not as much noise is removed, and the original audio signal stays relatively undisturbed.

Output Hiss Only Lets you preview only hiss to determine if the effect is removing any desirable audio.

Advanced settings Click the triangle to display these options:

Spectral Decay Rate When audio is encountered above the estimated noise floor, determines how much audio in surrounding frequencies is assumed to follow. With low values, less audio is assumed to follow, and hiss reduction will cut more closely to the frequencies being kept.

Values of 40% to 75% work best. If the value is too high (above 90%), unnaturally long tails and reverbs might be heard. If the value is too low, background bubbly effects might be heard, and music might sound artificial.

Precision Factor Determines the time-accuracy of hiss reduction. Typical values range from 7 to 14. Lower values might result in a few milliseconds of hiss before and after louder parts of audio. Larger values generally produce better results and slower processing speeds. Values over 20 don't ordinarily improve quality any further.

Transition Width Produces a slow transition in hiss reduction instead of an abrupt change. Values from 5 to 10 usually achieve good results. If the value is too high, some hiss may remain after processing. If the value is too low, background artifacts might be heard.

FFT Size Specifies a **Fast Fourier Transform** size, which determines the tradeoff between frequency- and time-accuracy. In general, sizes from 2048 to 8192 work best.

Lower FFT sizes (2048 and below) result in better time response (less swooshing before cymbal hits, for example), but they can produce poorer frequency resolution, creating hollow or flanged sounds.

Higher FFT sizes (8192 and above) might cause swooshing, reverb, and drawn out background tones, but they produce very accurate frequency resolution.

Control Points Specifies the number of points added to the graph when you click **Capture Noise Floor**.

Watch the Clean up background noise and reduce hiss to learn how to clean up background noises and apply hiss reduction to audio with Adobe Audition.

Finding and customizing shortcuts

Adobe Audition provides a default set of keyboard shortcuts to help you speed up the editing process. In menus and tool tips, available keyboard shortcuts appear to the right of command and button names. You can customize nearly all default shortcuts and add shortcuts for other functions.

Visual Keyboard Shortcut Editor

You can use the Visual Keyboard Shortcut editor to view the keys with their assigned shortcuts, assign new shortcuts, and delete existing ones. The feature also lets you easily navigate and provides maximum workflow efficiency. A tool tip reveals the full command name when you hover over a key in the Keyboard layout.

When you select a key on the Keyboard Layout, you can view all the commands that are assigned to that key and all other modifier combinations.

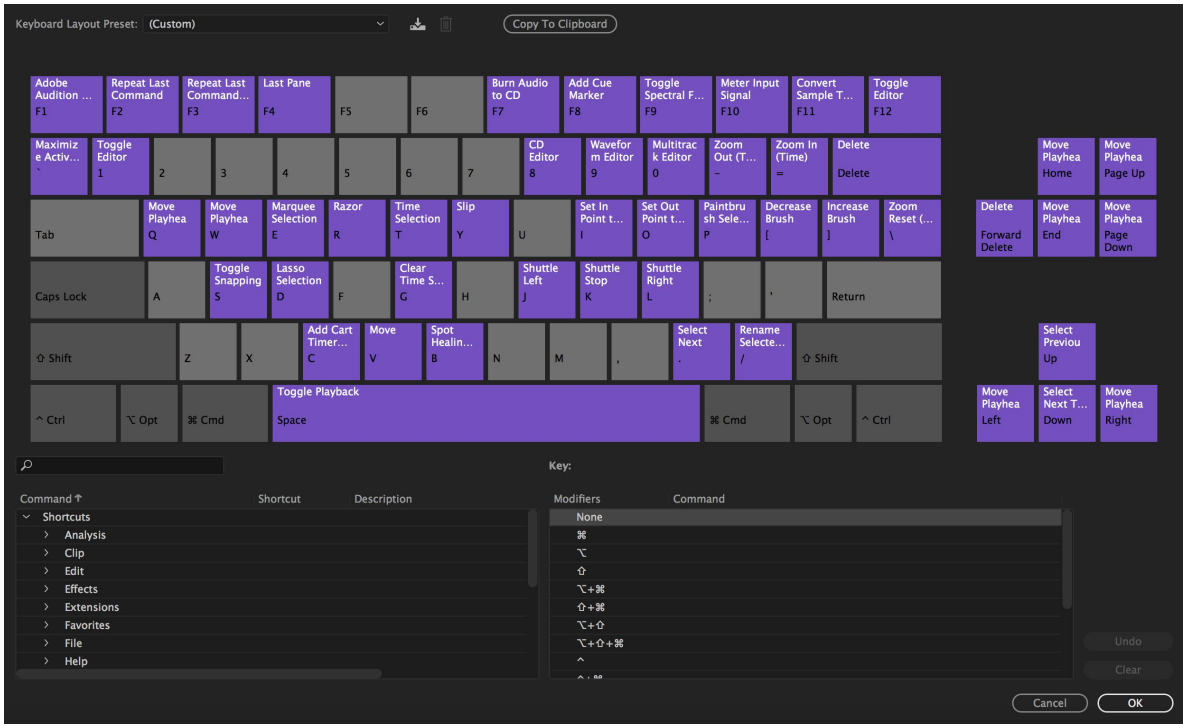
- Audition detects the keyboard hardware and the appropriate keyboard layout is displayed accordingly.
- When Audition detects a non-supported keyboard, the default view is to display the U.S. English keyboard. By default, the **Adobe Audition Default** preset is displayed.

- When you change a shortcut, the preset pop-up menu changes to **Custom**. After you make the required changes, use the **Save As** option to save the customized shortcut set as a preset.

Using and customizing the Visual Keyboard Shortcut

- To view the Visual Keyboard Shortcuts, select **Edit** < **Keyboard Shortcuts**.

The **Keyboard Shortcut** window appears which shows you the layout of a keyboard. The keys with shortcuts assigned are marked in Purple. All the shortcuts (marked in purple) are application-wide shortcuts.

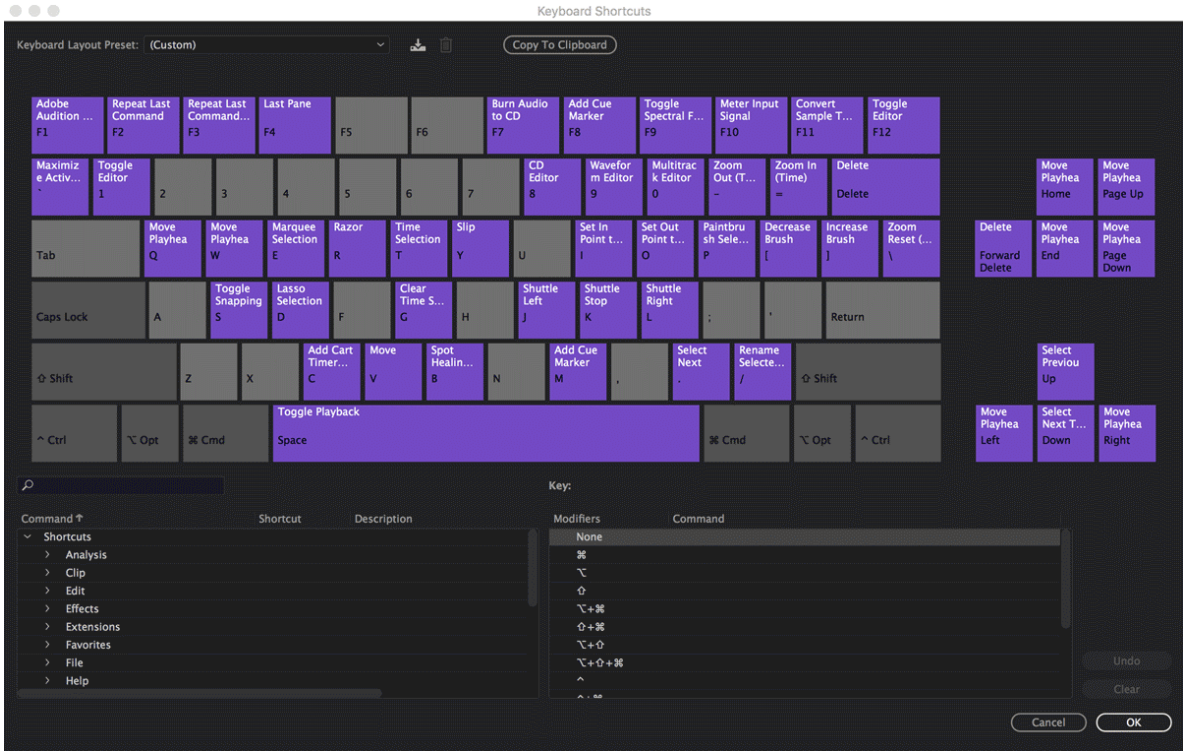


Visual Keyboard Shortcut Editor

Shortcut assignment using drag-and-drop

You can assign shortcuts by dragging a command from the **Command** list to a key in the Keyboard Layout. You can also apply shortcuts using the modifier combination for the currently selected key displayed in the Key Modifier List. The Command List and the **Modifier** list appear below the keyboard.

- To assign a command to a key along with a modifier, hold down the modifiers during drag-and-drop.
- To remove the last assigned shortcut command, select the key and click **Undo**.
- To remove all shortcuts assigned, click **Clear**.



Assign shortcut

Visual Shortcut Editor in Audition CC

Video

Conflict resolution

When an existing shortcut conflicts with another command:

- A warning appears at the bottom of the editor.
- Undo and Clear buttons in the lower right corner are enabled.
- The command in conflict is highlighted in blue, and clicking the key automatically selects the command in the command list.

You can now easily change the assignment for the conflicting command.

Avid ProTools shortcuts in Audition

You can apply default ProTools shortcuts in Audition. ProTools is a digital audio workstation that functions as a multitrack tape recorder and audio mixer along with more features.

Using ProTools shortcuts

To apply ProTools shortcuts in Audition, use the following steps.

1 Select **Edit** < **Keyboard Shortcuts.**

2 In the **Keyboard Shortcuts** window, select **Keyboard Layout Preset**.

3 From the drop-down menu, select **ProTools (read only)**.

You can now use ProTools shortcuts in Audition.

Find shortcuts

? Do any of the following:

- For menu commands, look for shortcuts to the right of command names.
- For tools or buttons, look for shortcuts to the right of tool tips. (To display tool tips, hold the pointer over a tool or button.)
- For a complete list of all shortcuts, choose Edit > Keyboard Shortcuts.

Customize shortcuts

You can customize nearly all default keyboard shortcuts and add shortcuts for other commands.

1 Choose Edit > Keyboard Shortcuts.

2 In the Command column, select the command you want to customize.

3 If you want to replace or remove an existing shortcut, choose it from the Shortcuts For Command menu.

4 Do any of the following:

- To create a shortcut, click inside the Press Shortcut box, and press the desired key combination. Then click Assign.
- To remove a shortcut, click Remove.

If you enter a key combination that's already in use, Audition displays an alert. Click Yes to transfer the shortcut to a different command, or No to retain the existing assignment.

Assign shortcuts to the Workspace commands to quickly switch between custom panel layouts.

Save or delete custom sets of shortcuts

1 Choose Edit > Keyboard Shortcuts.

2 Do either of the following:

- To save a custom set, click Save As, enter a name, and click OK.
- To delete a custom set, choose it from the Set menu, and then click Delete.

Restore the default shortcuts

1 Choose Edit > Keyboard Shortcuts.

2 From the Set menu, choose Default Set.

Adobe Audition manual (PDF)

Find a PDF of articles to learn how to use Adobe Audition.

Adobe Audition manual (PDF)

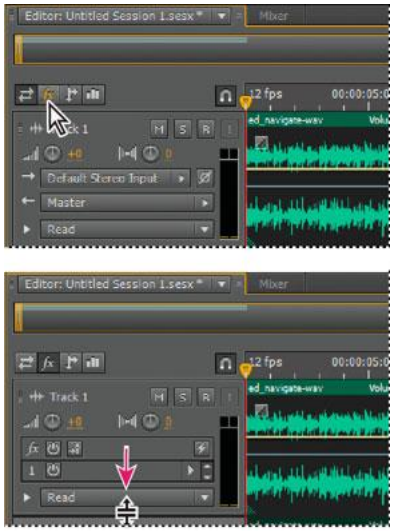
Applying effects in the Multitrack Editor

Apply effects to clips or tracks

In the Multitrack Editor, you can apply up to 16 effects to each clip, track, and bus and adjust them while a mix plays. (Apply clip effects if a track contains multiple clips that you want to process independently.)

You can insert, reorder, and remove effects in the Editor, Mixer, or Effects Rack panel. Only in the Effects Rack, however, can you save favorite settings as presets, which you can apply to multiple tracks.

In the Multitrack Editor, effects are *nondestructive*, so you can change them at any time. To readapt a session for different projects, for example, simply reopen it and change effects to create new sonic textures.



Revealing effect slots in the Editor panel

1. Do any of the following:
   - Select a clip, and click Clip Effects at the top of the Effects Rack.
   - Select a track, and click Track Effects at the top of the Effects Rack.
   - Display the *fx* section of the Editor or Mixer. (In the Editor panel, click the button fx in the upper-left corner.)
2. Choose effects for up to 16 slots in the list. (See Insert, bypass, reorder, or remove effects in racks.)
3. Press the spacebar to play the session, and then edit, reorder, or remove effects as needed.

To change effect settings over time, use envelopes. (See Automating mixes with envelopes.)

Pre-render track effects to improve performance

In the Multitrack Editor, pre-render track effects to address heavy CPU usage, improving performance for complex mixes or low-latency recording. (Latency measures the delay between user input and sound output from a computer. If latency is high, it produces an audible echo during recording, disrupting timing for musicians.)

You can continue to edit track settings normally; pre-rendering processes audio when pauses occur in playback or editing.

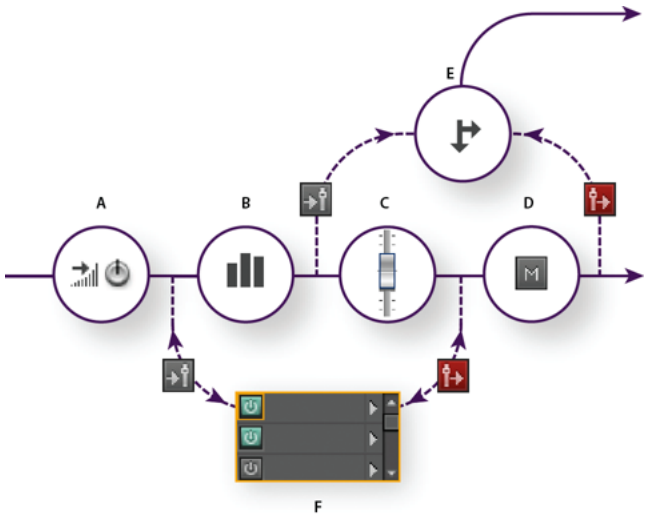
? In the Editor panel, Effects Rack, or Mixer, click the Pre-Render Track button .

Insert effects before or after sends and EQ

On each track, you can insert effects either pre- or post-fader. Pre-fader effects process audio *before* sends and EQ. Post-fader effects process audio *after* sends and EQ. For most mixes, the default, pre-fader setting works well. The post-fader setting offers signal-routing flexibility for particularly complex mixes.

? In the fx section of the Editor panel or Mixer, click the Pre-Fader/Post-Fader button to insert effects either before sends and EQ , or after .

If you're editing effect settings in the Effects Rack, click the Pre-Fader/Post-Fader button in the lower-left corner.



Pre- and post-fader effect and send routing for each track:
***A** Input **B** EQ **C** Volume **D** Mute **E** Send **F** Effects Rack*

Chapter 2: Workspace and setup

Control surface support

Support for PreSonus FaderPort

The PreSonus FaderPort is a hardware control surface device that connects to your computer via USB, delivering transport and mixing control. You can use the device to start and stop audio playback and recording, adjust volume with the motorized fader, control pan, and automation. All functions happen in real time along with other functions.

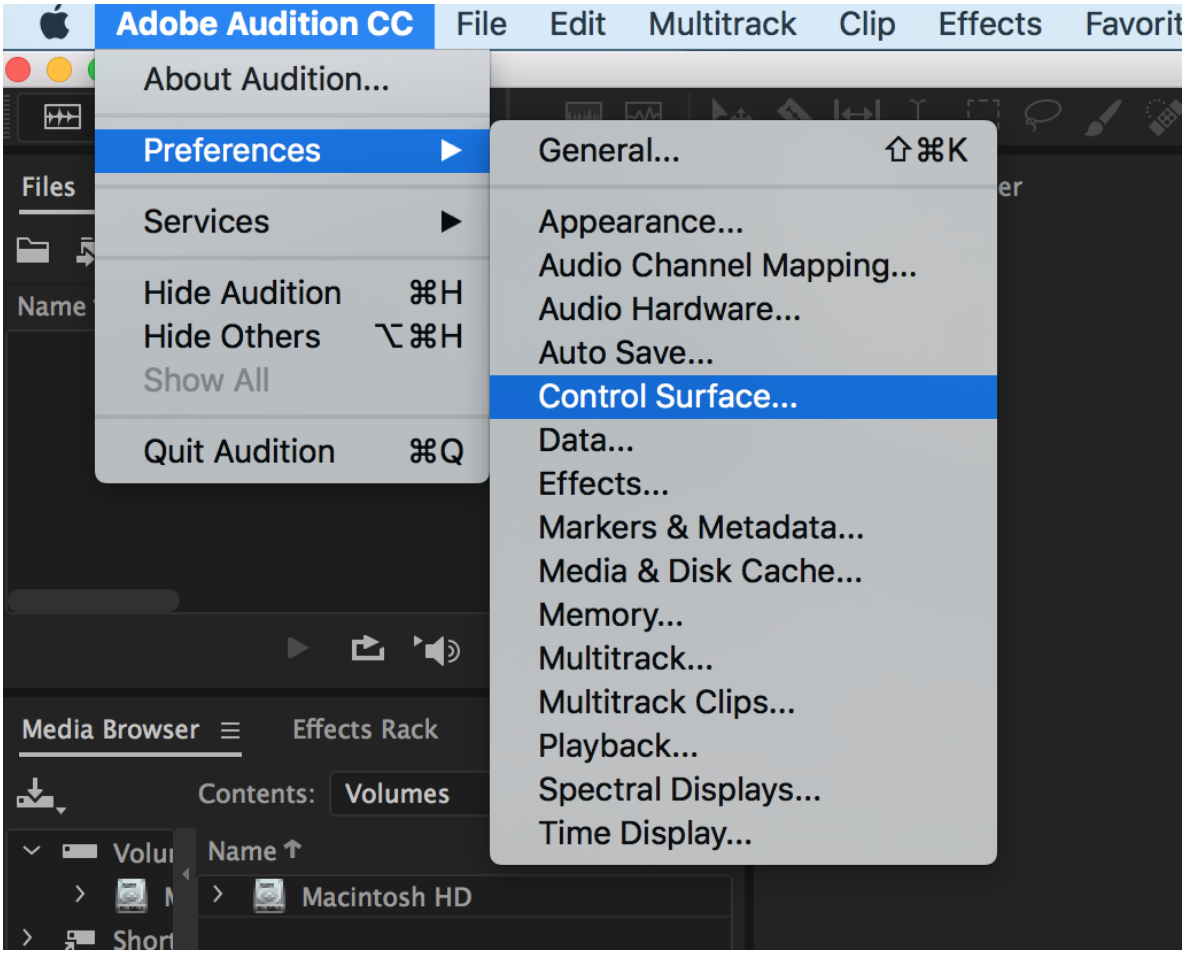
For more information, see PreSonus FaderPort.

Note:

Note: Audition does not support the FaderPort 8 device.

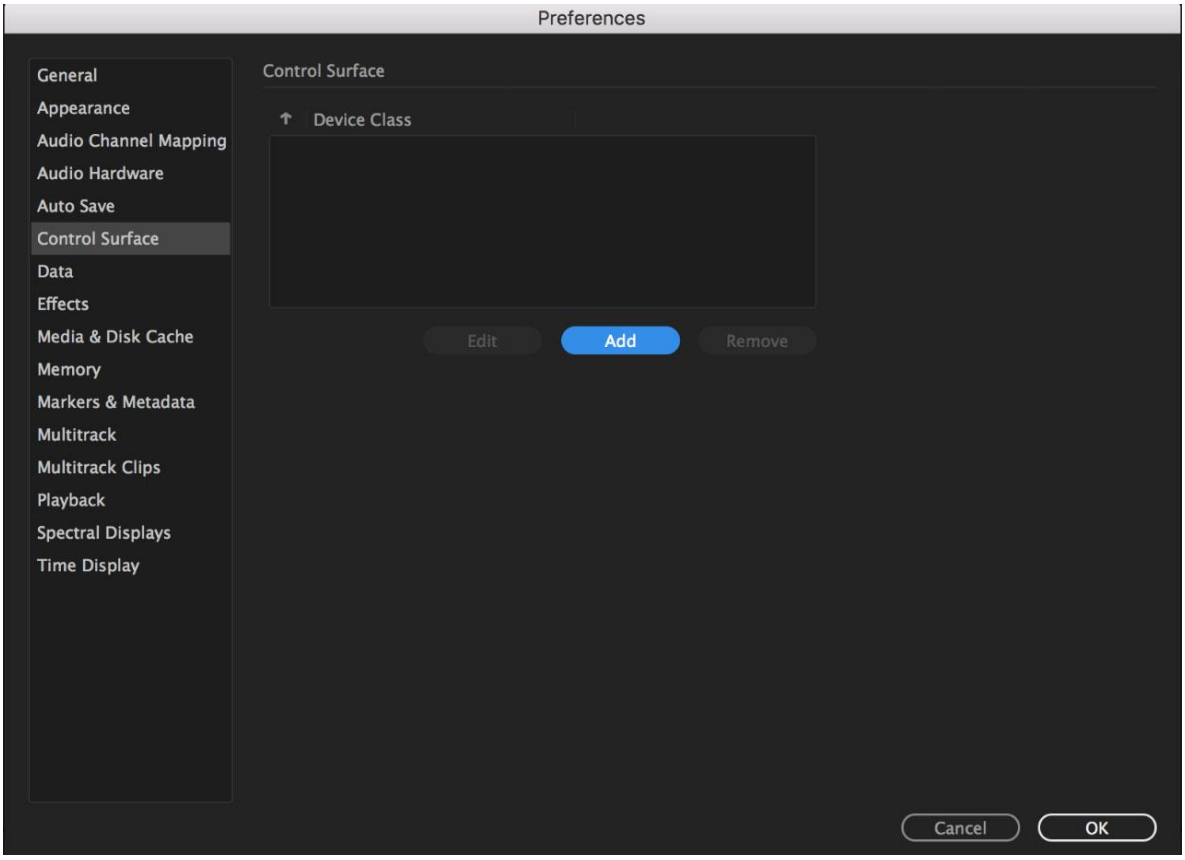
To use FaderPort with Audition, follow these steps:

1 Select **Edit** > **Preferences** > **Control Surface** (Windows) or **Adobe Audition CC** > **Preferences** > **Control Surface** (MAC).



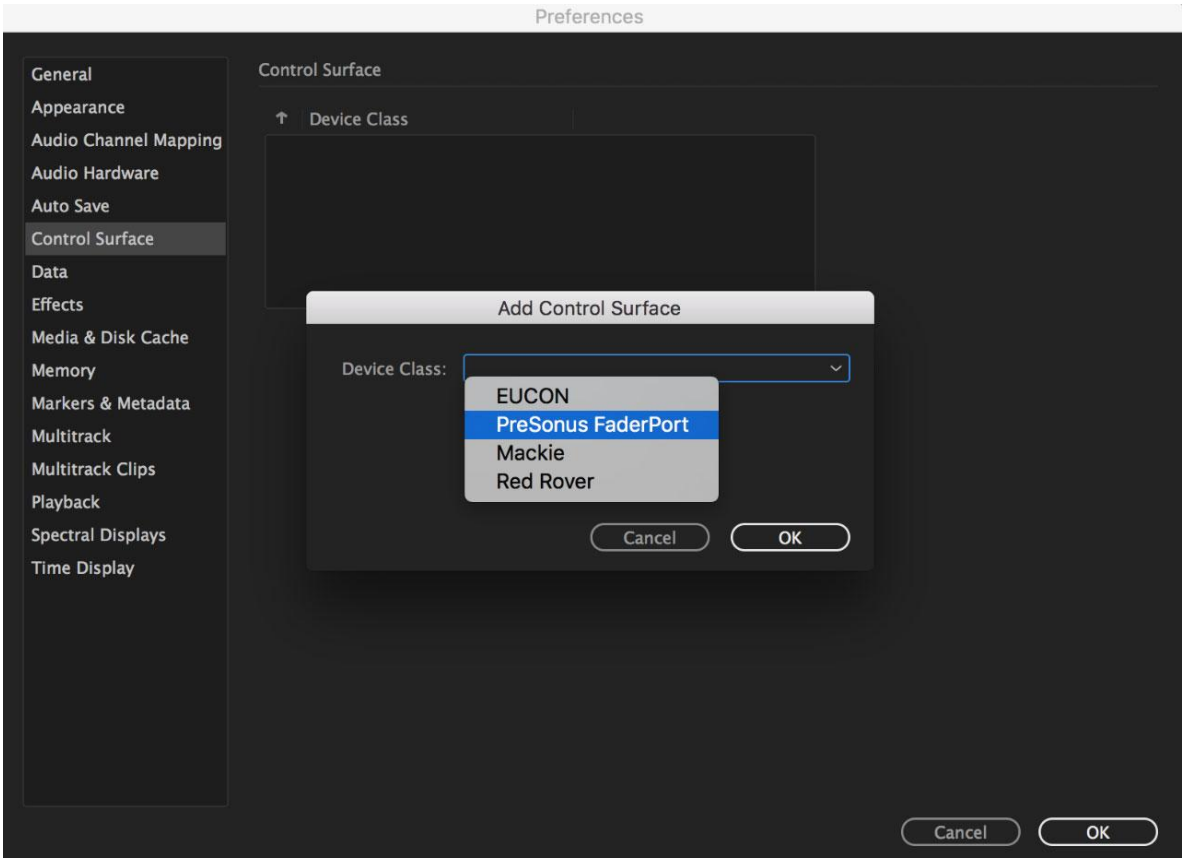
Preference setting

2 In the Preferences dialog box, click **Add**.



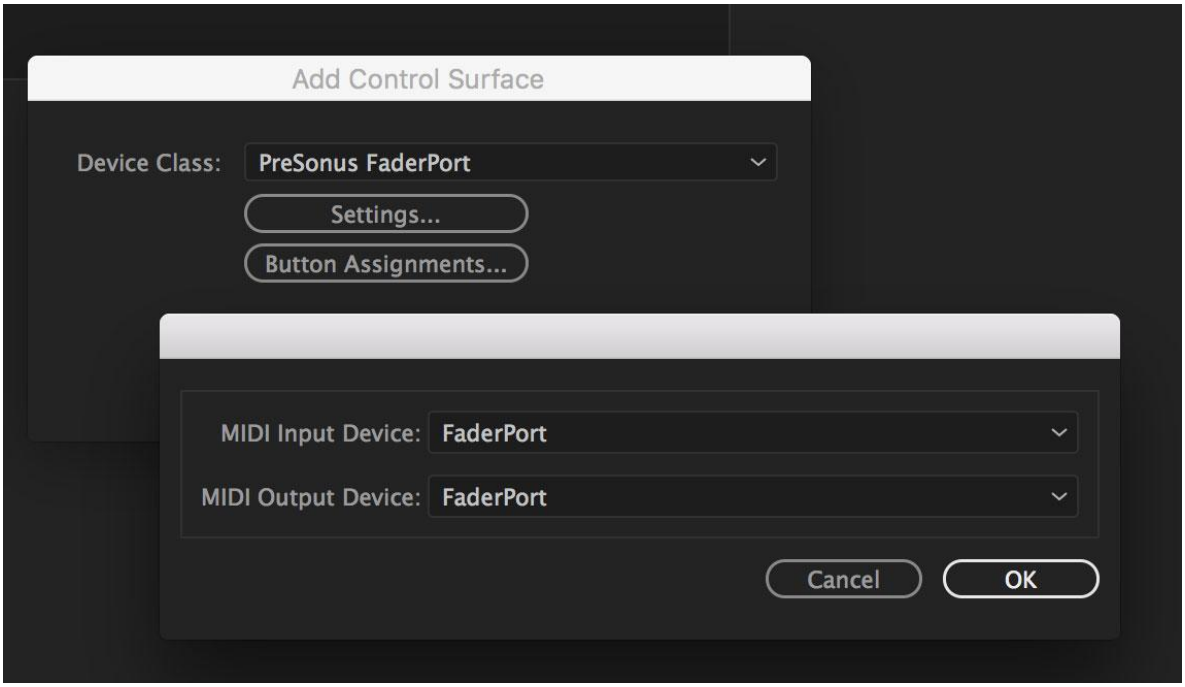
Add Preferences

3 In the **Add Control Surface** dialog box, select **PreSonus FaderPort**.



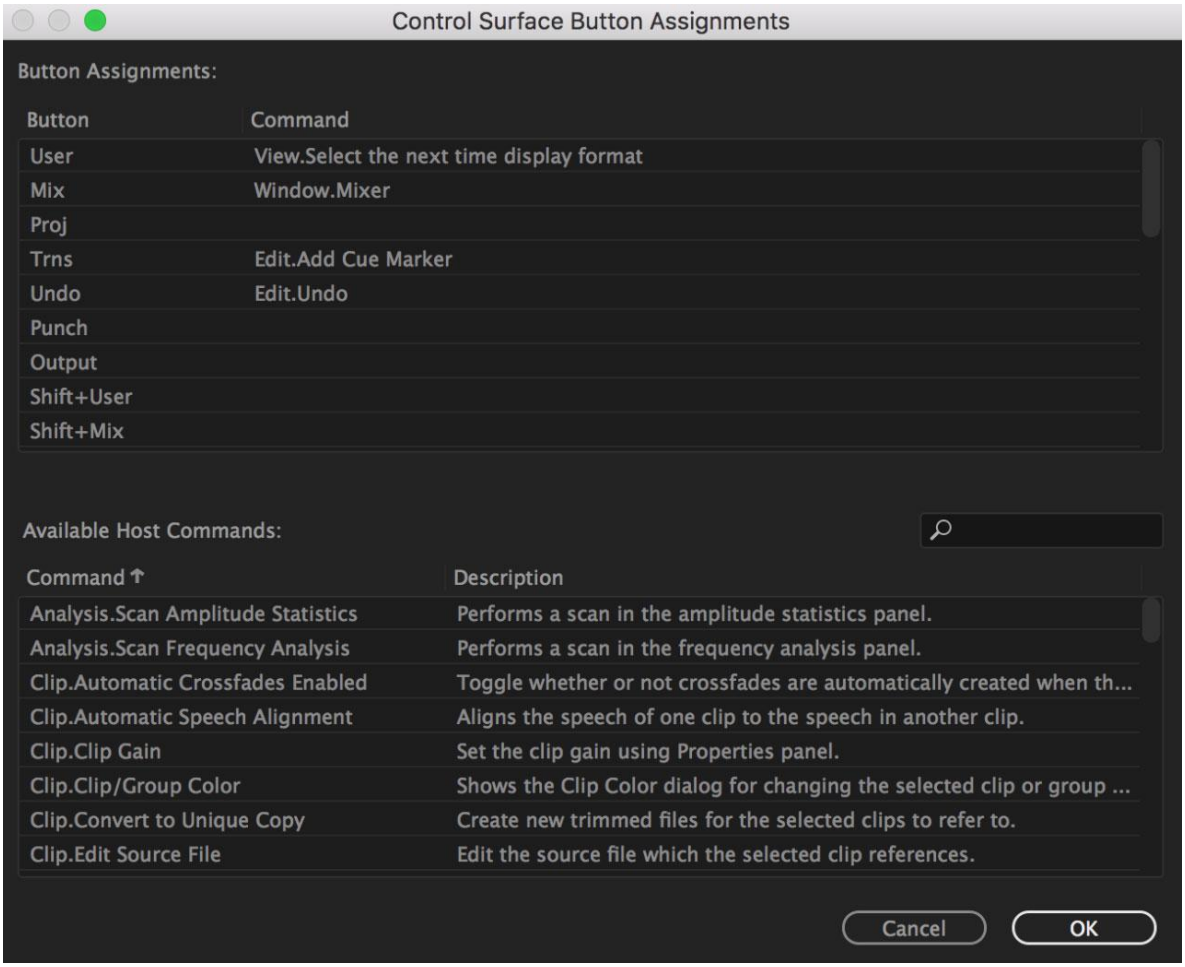
Add control surface

4 To configure MIDI input devices and MIDI output devices, click **Settings**.



Control surface settings

5 To edit button assignment settings, click **Button Assignments**.



Button Assignments setting

6 Click **OK**.

PreSonus FaderPort is now ready to be used with Audition.

Support for Mackie HUI

You can use HUI communications protocol to interface between the audio control surface and Adobe Audition. The protocol allows Audition and a connected hardware control surface to synchronize the states of their faders, buttons, jog wheels, and displays.

Note:

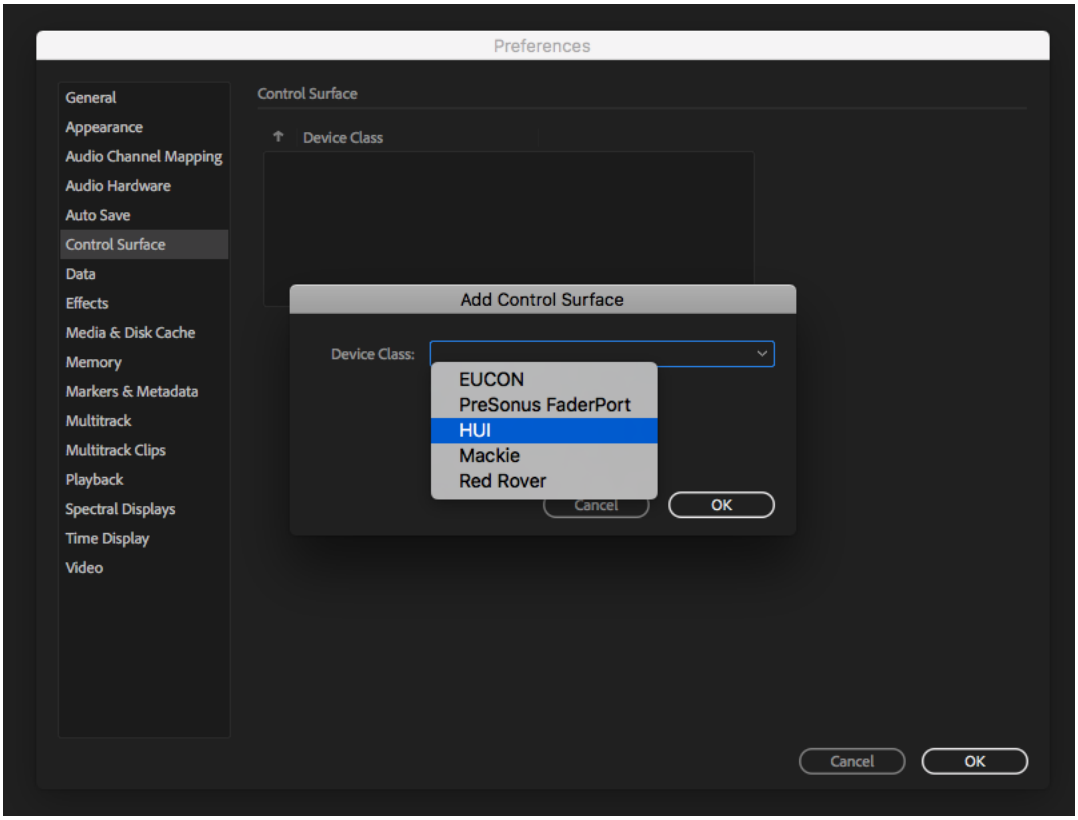
Note: Support for control surfaces that emulate HUI depends on the device's implementation of the protocol.

To connect your HUI-compatible device with Audition, follow these steps:

1 Select **Edit** > **Preferences** > **Control Surface** (Windows) or **Adobe Audition CC** > **Preferences** > **Control Surface** (MAC).

2 In the Preferences dialog box, click **Add**.

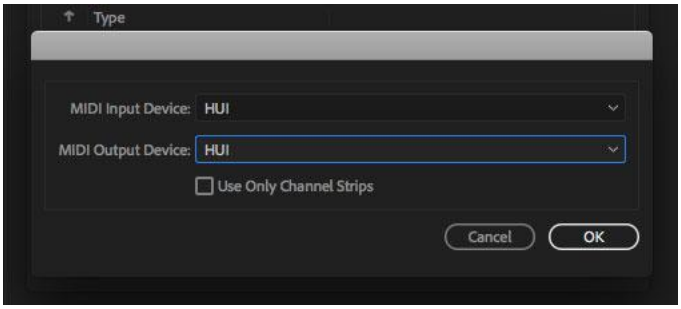
3 In the **Add Control Surface** dialog box, select **HUI**.



Add control surface

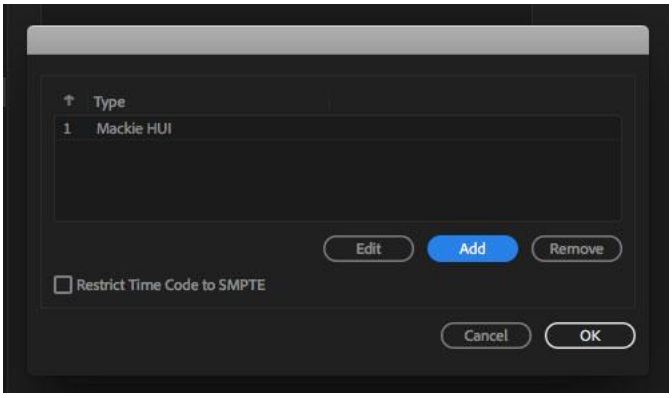
4 To adjust settings, click **Settings**.

5 To restrict control surface controls to the control strips, enable the **Use Only Channel Strips** check box. To allow assignment to application commands via the **Control Surface Button Assignments** dialog, the user-defined buttons are still enabled for the device.



Configure settings

6 If you want to display SMPTE frame rates only as the time base displayed on the device, enable the **Restrict Time Code to SMPTE** check box. If this check box is selected, Bar and Beats, Decimal, Compact Disc, and Custom is not displayed on the control surface.



Restrict Time Code

7 To edit button assignments, click **Button Assignments** in the **Add Control Surface** dialog box.

8 Click **OK.**

Viewing, zooming, and navigating audio

Comparing the Waveform and Multitrack editors

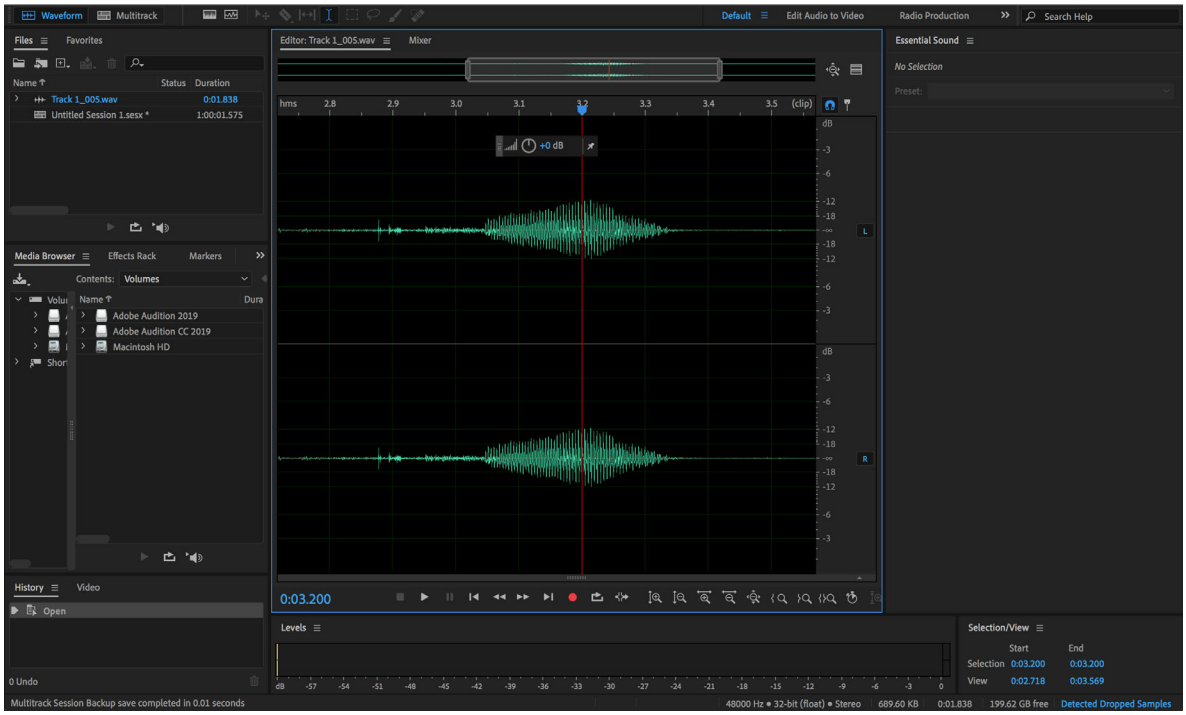
Adobe Audition provides different views for editing audio files and creating multitrack mixes. To edit individual files, use the Waveform Editor. To mix multiple files and integrate them with video, use the Multitrack Editor.

The Waveform and Multitrack editors use different editing methods, and each has unique advantages. The Waveform Editor uses a destructive method, which changes audio data, permanently altering saved files. Such permanent changes are preferable when converting sample rate and bit depth, mastering, or batch processing. The Multitrack Editor uses a nondestructive method, which is impermanent and instantaneous, requiring more processing power, but increasing flexibility. This flexibility is preferable when gradually building and reevaluating a multilayered musical composition or video soundtrack.

You can combine destructive and nondestructive editing to suit the needs of a project. If a multitrack clip requires destructive editing, for example, simply double-click it to enter the Waveform Editor. Likewise, if an edited waveform contains recent changes that you dislike, use the Undo command to revert to previous states—destructive edits aren't applied until you save a file.

Basic components of the editors

Though available options differ in the Waveform and Multitrack editors, both views share basic components, such as the tool and status bars, and the Editor panel.



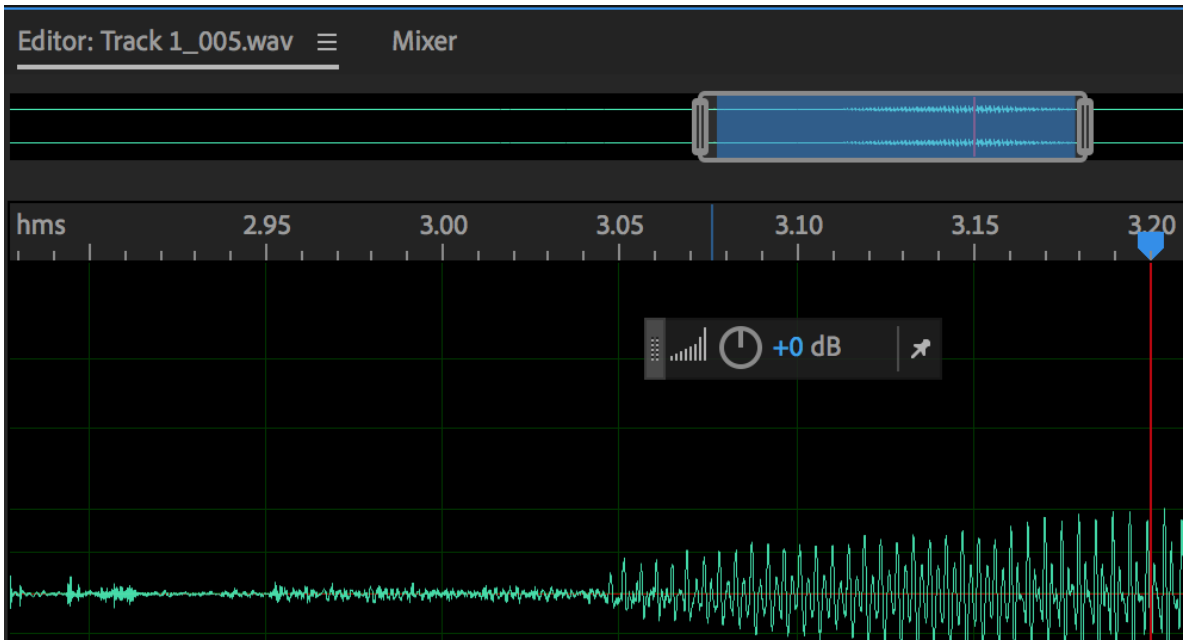
Basic Waveform view

Switch editors

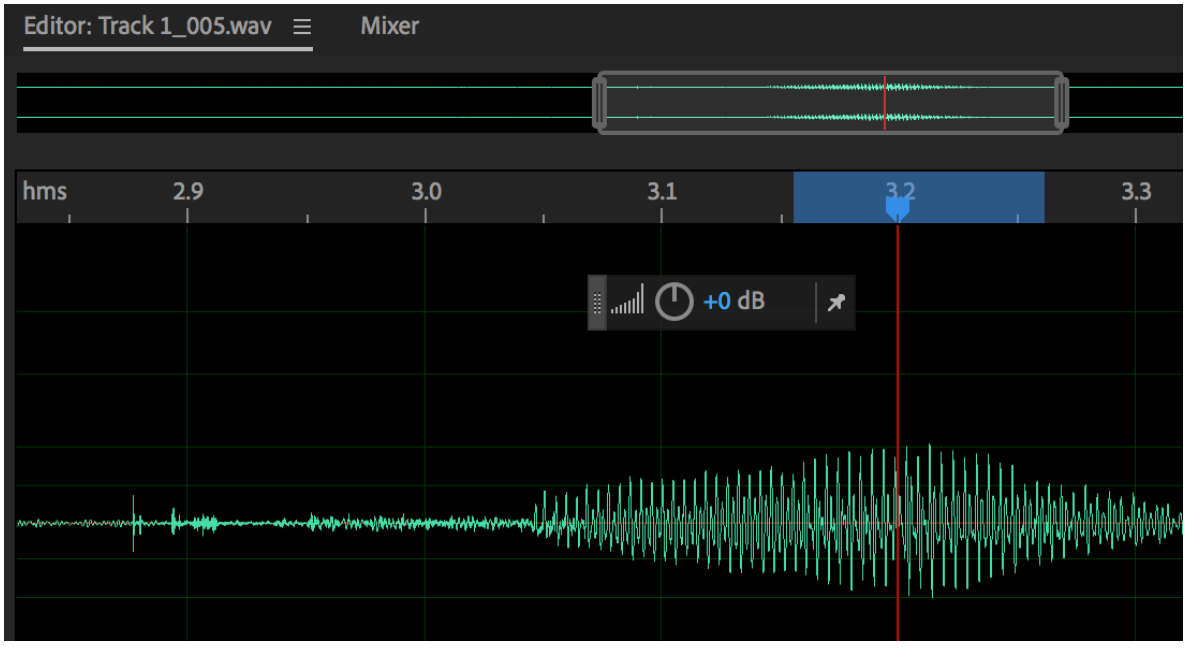
? Do one of the following:

- From the View menu, choose Waveform or Multitrack Editor.
- In the toolbar, click the Waveform or Multitrack Editor button.
- In the Multitrack Editor, double-click an audio clip to open it in the Waveform Editor. Alternatively, double-click a file in the Files panel.
- In the Waveform Editor, choose Edit > Edit Original to open the multitrack session that created a mixdown file. (This command requires embedded metadata in the file. See Link sessions to exported mixdown files.)

Zoom audio in the Editor panel



Zoom navigator



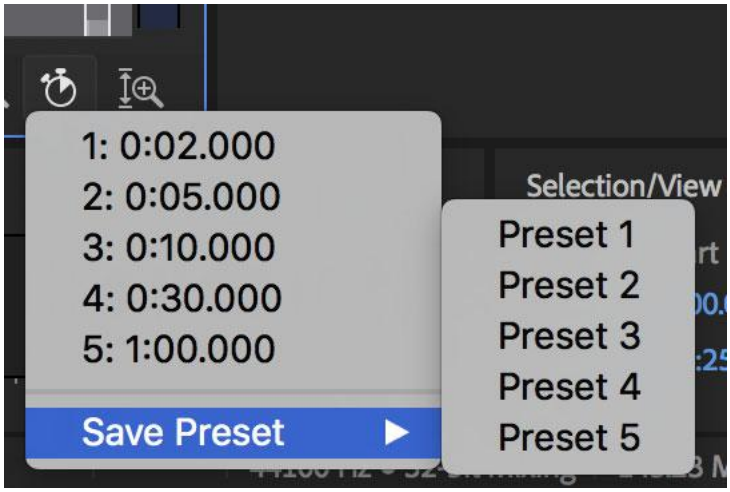
Timeline ruler

Zoom into a specific time range

In either the zoom navigator or the timeline ruler, right-click and drag. The magnifying glass icon creates a selection showing the range that will fill the **Editor** panel.

Zoom to preset

Click on the Timer icon on the bottom of the **Editor** panel. There are five preset slots available that can be used to zoom to a specific time. In the **Save Preset** menu, you can overwrite a preset slot with the duration that is currently selected.



Zoom to time

Zoom to selected clip

Quickly zoom to the time range of one or more selected clips using the Zoom menu command. In the Multitrack Editor, choose **Zoom > Zoom To Selected Clips**.

Zoom into a specific frequency range

In the vertical ruler for the spectral display, right-click and drag. (See View audio waveforms and spectrums.)

Extend or shorten the displayed range

Place the pointer over the left or right edge of the highlighted area in the zoom navigator, and then drag the magnifying glass icon .

Gradually zoom in or out

In the lower right of the Editor panel, click the Zoom In or Zoom Out button.

You can set the Zoom Factor in the General section of the Preferences dialog box. (See Customize preferences.)

Zoom out full (all tracks)

You can zoom out all tracks to the same height to fully cover vertical spaces. The view will resize track heights to take up the full height of the multitrack editor panel. Track heights will resize to a consistent height. Minimized tracks will still remain at their minimum height.

To Zoom out full, choose **View > Zoom Out Full (All Tracks)**.

Zoom with the mouse wheel or Mac trackpad

Place the pointer over the zoom navigator or ruler, and either roll the wheel or drag up or down with two fingers. (In the Waveform Editor, this zoom method also works when the pointer is over the waveform.)

Roll or drag over the spectral display, and press Shift to switch between logarithmic and linear frequency scales. (Logarithmic better reflects human hearing; linear makes individual frequencies more visually distinct.)

Magnify selected audio

In the lower right of the Editor panel, click the Zoom In At In Point , Zoom In At Out Point , or Zoom To Selection buttons.

Display the entire audio file or multitrack session

In the lower right of the Editor panel, click the Zoom Out Full button .

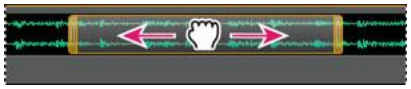
To display zoom buttons in a separate panel, choose Window > Zoom.

Navigate through time

At higher zoom levels, you can navigate to different audio content in the Editor panel.

Navigate by scrolling

- In the zoom navigator, drag left or right.



Scrolling with the zoom navigator

- Use horizontal mouse scroll or a middle button mouse drag to scroll the timeline in multitrack and waveform view.
- To scroll through audio frequencies in the spectral display, drag up or down in the vertical ruler. (See View audio waveforms and spectrums.)

Navigate with the Selection/View panel

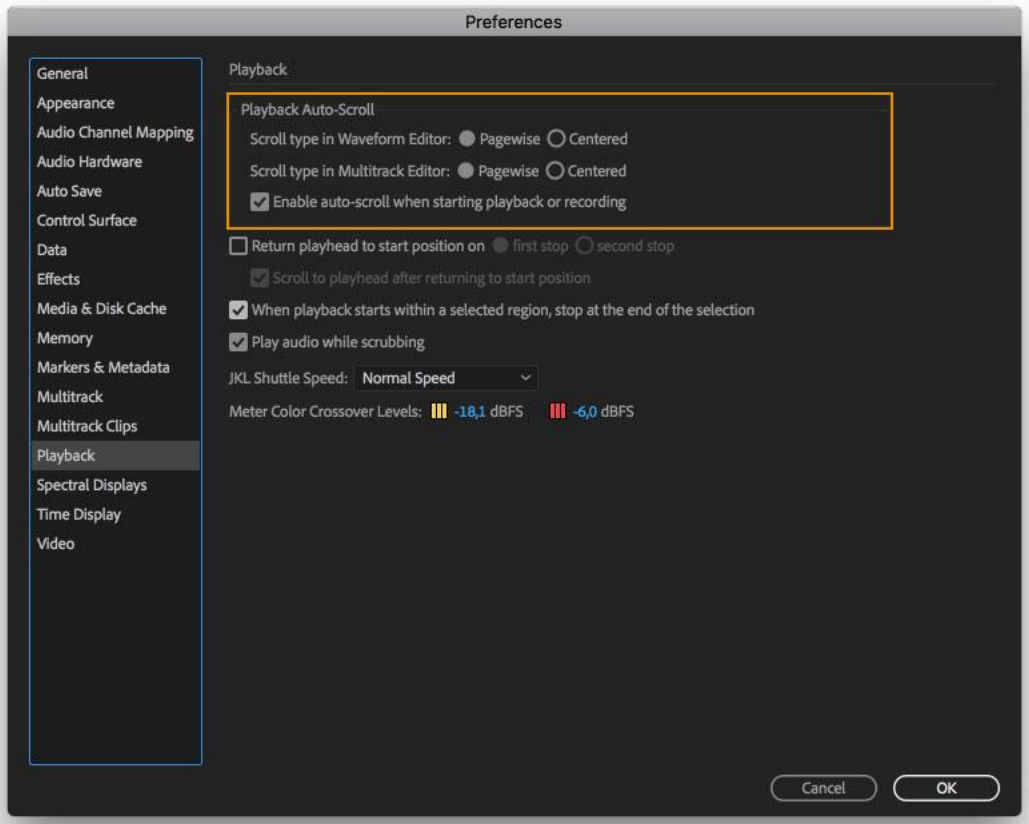
The Selection/View panel shows the start and end of the current selection and view in the Editor panel. The panel displays this information in the current time format, such as Decimal or Bars And Beats. (See Change the time display format.)

1. To display the Selection/View panel, choose Window > Selection/View Controls.
2. (Optional) Enter new values into the Begin, End, or Duration boxes to change the selection or view.

Auto-scroll navigation

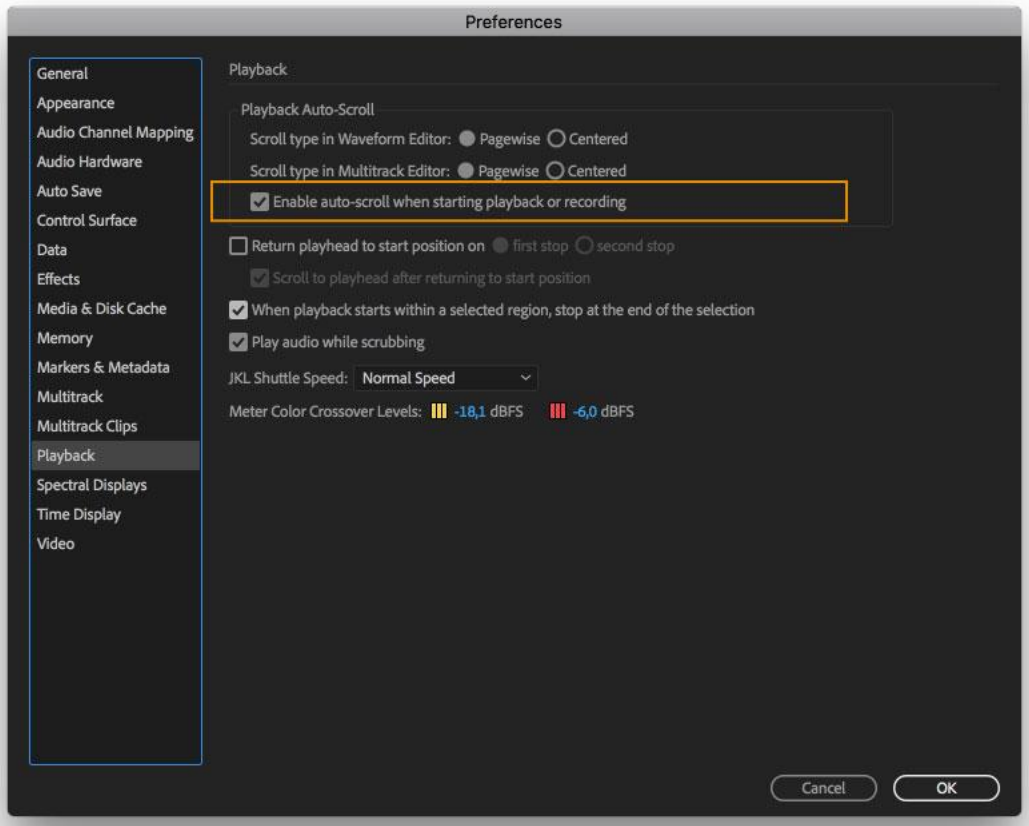
You can use auto-scroll to navigate on the waveform and multitrack editor. To choose the scroll type, open **Preferences > Playback**. Use the radio buttons to choose the type of scroll individually for both the editors.

- Pagewise scroll: The playhead moves from left to right and jumps to the next frame when it hits the right corner.
- Centered scroll: The playhead is positioned at the center and the track beneath it moves. Therefore, the current time of audio being played is always in the middle.



Auto-scroll under playback preferences

To set your preference to **Enable auto-scroll when starting playback or recording**, enable the checkbox.



Enable auto-scroll on start

Use the auto-scroll indicator ( ) available in the **Editor** panel on the top-right corner in the Waveform editor, and top-left corner in the Multitrack editor, to enable or disable auto-scroll. Note that manually moving the playhead when auto-scroll is enabled, automatically disables the function.

Connecting to audio hardware in Audition

You can use a wide range of hardware inputs and outputs with Adobe Audition. Sound card inputs let you bring in audio from sources such as microphones, tape decks, and digital effects units. Sound card outputs let you monitor audio through sources such as speakers and headphones.

***A** Sound card inputs connect to sources such as microphones and tape decks. **B** Sound card outputs connect to speakers and headphones.*

Configure audio inputs and outputs

When you configure inputs and outputs for recording and playback, Adobe Audition can use these kinds of sound card drivers:

- In Windows, ASIO drivers support professional cards and MME drivers typically support standard cards.
- In Mac OS, CoreAudio drivers support both professional and standard cards.

ASIO and CoreAudio drivers are preferable because they provide better performance and lower latency. You can also monitor audio as you record it and instantly hear volume, pan, and effects changes during playback.

1. Choose Edit > Preferences > Audio Hardware (Windows) or Audition > Preferences > Audio Hardware (Mac OS).
2. From the Device Class menu, choose the driver for the sound card you want to use.
3. Choose a Default Input and Output from the card.

   In the Multitrack Editor, you can override the defaults for specific tracks. See Assign audio inputs and outputs to tracks.
4. (MME and CoreAudio) For Master Clock, choose the input or output to which you want other digital audio hardware to synchronize (ensuring accurate alignment of samples).
5. For I/O Buffer Size (ASIO and CoreAudio) or Latency (MME), specify the lowest setting possible without audio dropouts. The ideal setting depends on the speed of your system, so some experimentation may be necessary.
6. Choose a Sample Rate for the audio hardware. (For common rates for different output mediums, see Understanding sample rate.)
7. (Optional) To optimize the performance of ASIO and CoreAudio cards, click Settings. For more information, consult the documentation for the sound card.

Note:

By default, Adobe Audition controls ASIO sound cards while playing or monitoring audio. If you want to access the card in another application, select Release ASIO Driver In Background. (Audition still controls the card while recording to avoid having recordings suddenly stop.)

Apply machine-specific hardware settings for network users

In network environments, Audition preferences are stored with each user account. For editing, interface, and other preferences, this creates a customized experience for each user. Audio hardware preferences, however, should typically remain consistent on a given machine, ensuring that inputs and outputs on the installed audio interface are available in Audition.

1 Choose Edit > Preferences > Audio Hardware (Windows) or Audition > Preferences > Audio Hardware (Mac OS).

2 At the bottom of the Audio Hardware settings, select Use Machine-Specific Device Defaults. (Deselect this option only if users move an assigned audio interface from machine to machine.)

To instead duplicate hardware settings from one machine to others, search for and copy the MachineSpecificSettings.xml file.

Assign file channels to inputs and outputs

1 Choose Edit > Preferences > Audio Channel Mapping (Windows) or Audition > Preferences > Audio Channel Mapping (Mac OS).

2 To the far right of items in the Input and Output lists, click the triangles to choose a hardware port for each file channel.

This procedure also sets default outputs for the Master track in the Multitrack Editor. To override the defaults, see Assign audio inputs and outputs to tracks.

Customizing and saving application settings

Customize preferences

The Preferences dialog box lets you customize Adobe Audition's display, editing behavior, use of hard disk space, and other settings.

? Choose Edit > Preferences (Windows) or Audition > Preferences (Mac OS). Then choose the area you want to customize.

For information about a particular option, hover the mouse over it until a tooltip appears.

In the Media & Disk Cache preferences, choose your fastest drive for the Primary Temp folder, and a separate drive for the Secondary Temp folder. Select Save Peak Files to store information about how to display WAV files. (Without peak files, larger WAV files reopen more slowly.)

Restore preferences to default settings

Unexpected behavior may indicate damaged preferences files. To re-create preferences files, do the following.

? Hold down the Shift key, and start Adobe Audition.

Export and import customized application settings

Application settings files store all current preferences, effect settings, and workspaces. Export and import these files to store groups of customized settings for specific workflows, or transfer favorite settings to another machine.

1 Choose File > Export > Application Settings. Then specify a filename and location.

2 To reapply the settings at a later time, choose File > Import > Application Settings.

To import preferences from Audition 2.0 or 3.0, search your system for the audition_settings.xml file. You can import that file into both the Mac and Windows versions of Audition CS.

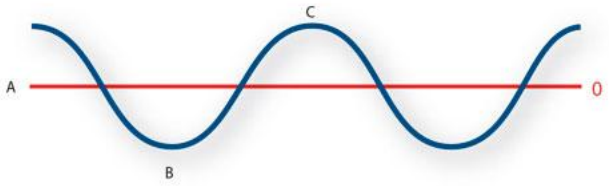
Chapter 3: Digital audio fundamentals

Understanding sound

Sound waves

Sound starts with vibrations in the air, like those produced by guitar strings, vocal cords, or speaker cones. These vibrations push nearby air molecules together, raising the air pressure slightly. The air molecules under pressure then push on the air molecules surrounding them, which push on the next set of molecules, and so on. As high-pressure areas move through the air, they leave low-pressure areas behind them. When these waves of pressure changes reach us, they vibrate the receptors in our ears, and we hear the vibrations as sound.

When you see a visual waveform that represents audio, it reflects these waves of air pressure. The zero line in the waveform is the pressure of air at rest. When the line swings up to a peak, it represents higher pressure; when the line swings down to a trough, it represents lower pressure.



A sound wave represented as a visual waveform
***A** Zero line **B** Low-pressure area **C** High-pressure area*

Waveform measurements

Several measurements describe waveforms:

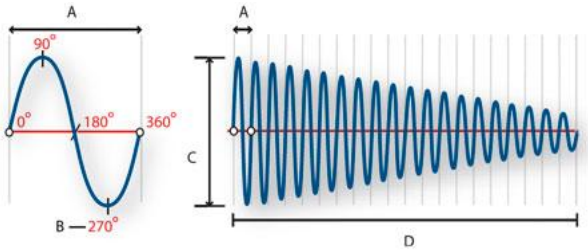
Amplitude Reflects the change in pressure from the peak of the waveform to the trough. High-amplitude waveforms are loud; low-amplitude waveforms are quiet.

Cycle Describes a single, repeated sequence of pressure changes, from zero pressure, to high pressure, to low pressure, and back to zero.

Frequency Measured in hertz (Hz), describes the number of cycles per second. (For example, a 1000-Hz waveform has 1000 cycles per second.) The higher the frequency, the higher the musical pitch.

Phase Measured in 360 degrees, indicates the position of a waveform in a cycle. Zero degrees is the start point, followed by 90° at high pressure, 180° at the halfway point, 270° at low pressure, and 360° at the end point.

Wavelength Measured in units such as inches or centimeters, is the distance between two points with the same degree of phase. As frequency increases, wavelength decreases.



A single cycle at left; a complete, 20-Hz waveform at right
***A** Wavelength **B** Degree of phase **C** Amplitude **D** One second*

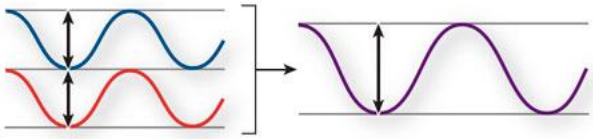
How sound waves interact

When two or more sound waves meet, they add to and subtract from each other. If their peaks and troughs are perfectly in phase, they reinforce each other, resulting in a waveform that has higher amplitude than either individual waveform.

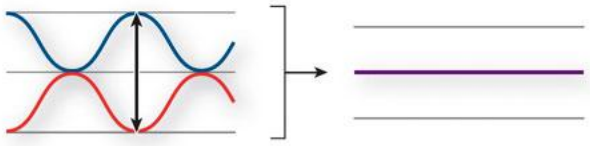
If the peaks and troughs of two waveforms are perfectly out of phase, they cancel each other out, resulting in no waveform at all.

In most cases, however, waves are out of phase in varying amounts, resulting in a combined waveform that is more complex than individual waveforms. A complex waveform that represents music, voice, noise, and other sounds, for example, combines the waveforms from each sound.

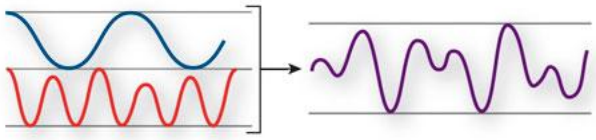
Because of its unique physical structure, a single instrument can create extremely complex waves. That's why a violin and a trumpet sound different even when playing the same note.



In-phase waves reinforce each other.



Out-of-phase waves cancel each other out.



Two simple waves combine to create a complex wave.

Digitizing audio

Comparing analog and digital audio

In analog and digital audio, sound is transmitted and stored very differently.

Analog audio: positive and negative voltage

A microphone converts the pressure waves of sound into voltage changes in a wire: high pressure becomes positive voltage, and low pressure becomes negative voltage. When these voltage changes travel down a microphone wire, they can be recorded onto tape as changes in magnetic strength or onto vinyl records as changes in groove size. A speaker works like a microphone in reverse, taking the voltage signals from an audio recording and vibrating to re-create the pressure wave.

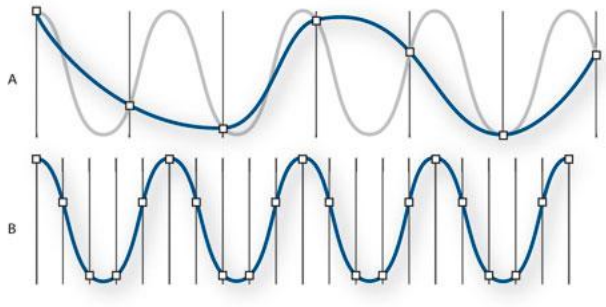
Digital audio: zeroes and ones

Unlike analog storage media such as magnetic tape or vinyl records, computers store audio information digitally as a series of zeroes and ones. In digital storage, the original waveform is broken up into individual snapshots called *samples*. This process is typically known as *digitizing* or *sampling* the audio, but it is sometimes called *analog-to-digital conversion*.

When you record from a microphone into a computer, for example, analog-to-digital converters transform the analog signal into digital samples that computers can store and process.

Understanding sample rate

Sample rate indicates the number of digital snapshots taken of an audio signal each second. This rate determines the frequency range of an audio file. The higher the sample rate, the closer the shape of the digital waveform is to that of the original analog waveform. Low sample rates limit the range of frequencies that can be recorded, which can result in a recording that poorly represents the original sound.



Two sample rates
***A** Low sample rate that distorts the original sound wave.* ***B** High sample rate that perfectly reproduces the original sound wave.*

To reproduce a given frequency, the sample rate must be at least twice that frequency. For example, CDs have a sample rate of 44,100 samples per second, so they can reproduce frequencies up to 22,050 Hz, which is just beyond the limit of human hearing, 20,000 Hz.

Here are the most common sample rates for digital audio:

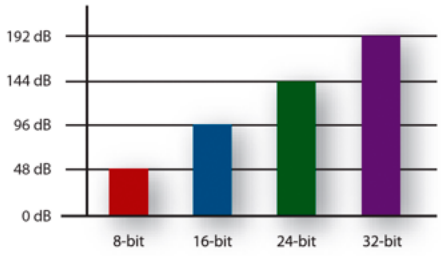
Sample rate	Quality level	Frequency range
11,025 Hz	Poor AM radio (low-end multimedia)	0–5,512 Hz
22,050 Hz	Near FM radio (high-end multimedia)	0–11,025 Hz
32,000 Hz	Better than FM radio (standard broadcast rate)	0–16,000 Hz
44,100 Hz	CD	0–22,050 Hz
48,000 Hz	Standard DVD	0–24,000 Hz
96,000 Hz	Blu-ray DVD	0–48,000 Hz

Understanding bit depth

Bit depth determines dynamic range. When a sound wave is sampled, each sample is assigned the amplitude value closest to the original wave's amplitude. Higher bit depth provides more possible amplitude values, producing greater dynamic range, a lower noise floor, and higher fidelity.

For the best audio quality, Audition transforms all audio in 32-bit mode and then converts to a specified bit depth when saving files.

Bit depth	Quality level	Amplitude values	Dynamic range
8-bit	Telephony	256	48 dB
16-bit	Audio CD	65,536	96 dB
24-bit	Audio DVD	16,777,216	144 dB
32-bit	Best	4,294,967,296	192 dB



Higher bit depths provide greater dynamic range.

Measuring amplitude in dBFS

In digital audio, amplitude is measured in decibels below full scale, or *dBFS*. The maximum possible amplitude is 0 dBFS; all amplitudes below that are expressed as negative numbers.

Note:

A given dBFS value does not directly correspond to the original sound pressure level measured in acoustic dB.

Audio file contents and size

An audio file on your hard drive, such as a WAV file, consists of a small header indicating sample rate and bit depth, and then a long series of numbers, one for each sample. These files can be very large. For example, at 44,100 samples per second and 16 bits per sample, a mono file requires 86 KB per second—about 5 MB per minute. That figure doubles to 10 MB per minute for a stereo file, which has two channels.

How Adobe Audition digitizes audio

When you record audio in Adobe Audition, the sound card starts the recording process and specifies what sample rate and bit depth to use. Through Line In or Microphone In ports, the sound card receives analog audio and digitally samples it at the specified rate. Adobe Audition stores each sample in sequence until you stop recording.

When you play a file in Adobe Audition, the process happens in reverse. Adobe Audition sends a series of digital samples to the sound card. The card reconstructs the original waveform and sends it as an analog signal through Line Out ports to your speakers.

To sum up, the process of digitizing audio starts with a pressure wave in the air. A microphone converts this pressure wave into voltage changes. A sound card converts these voltage changes into digital samples. After analog sound becomes digital audio, Adobe Audition can record, edit, process, and mix it—the possibilities are limited only by your imagination.

Chapter 4: Importing, recording, and playing

Multichannel audio workflow

Multichannel Audio Workflow

You can use Adobe Audition to easily work with multichannel audio up to 32 channels. In the **Waveform Editor**, you can individually edit and process specific audio channels. You can also work with multichannel files in the **Multitrack Editor** as a single clip or with the channels split across multiple clips.

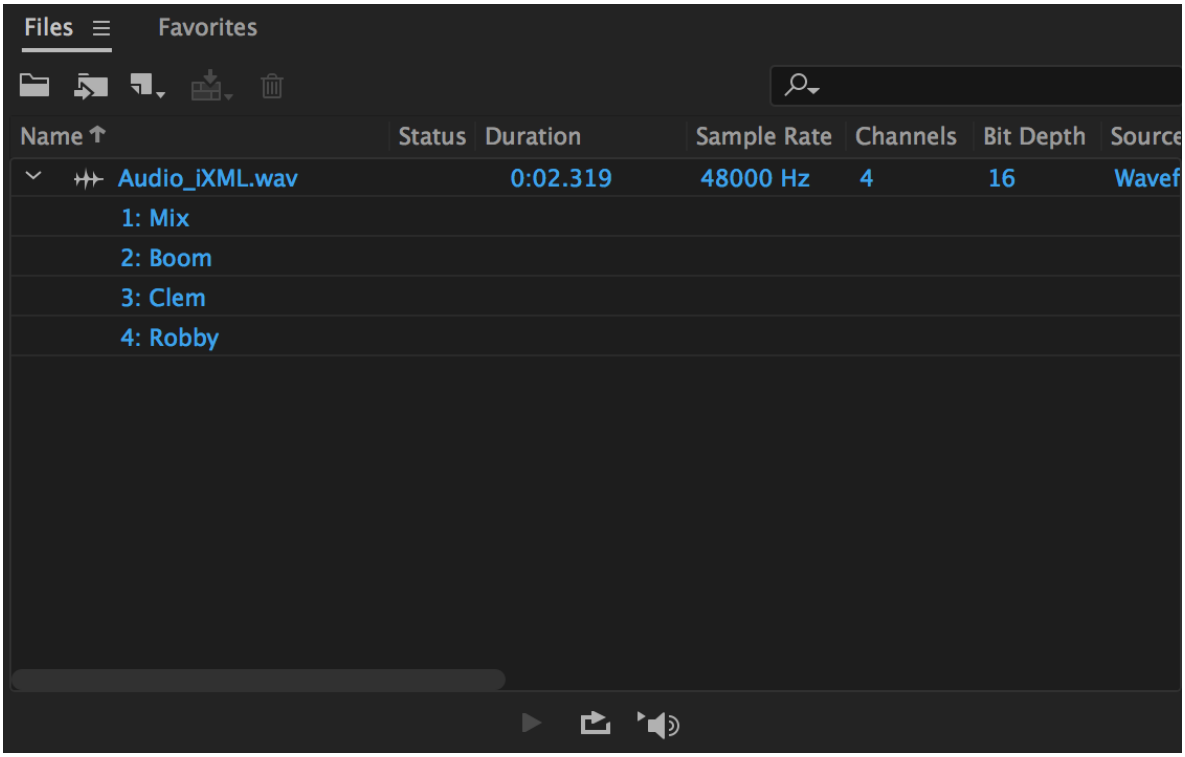
Video

Waveform Editor

If you have multichannel audio or video files, you can edit each audio channel separately in the **Waveform Editor** by following these steps.

1 Select **File** > **Open**. The opened file appears in the **Files Panel**.

2 Expand the drop-down to see each of the channels within the file.

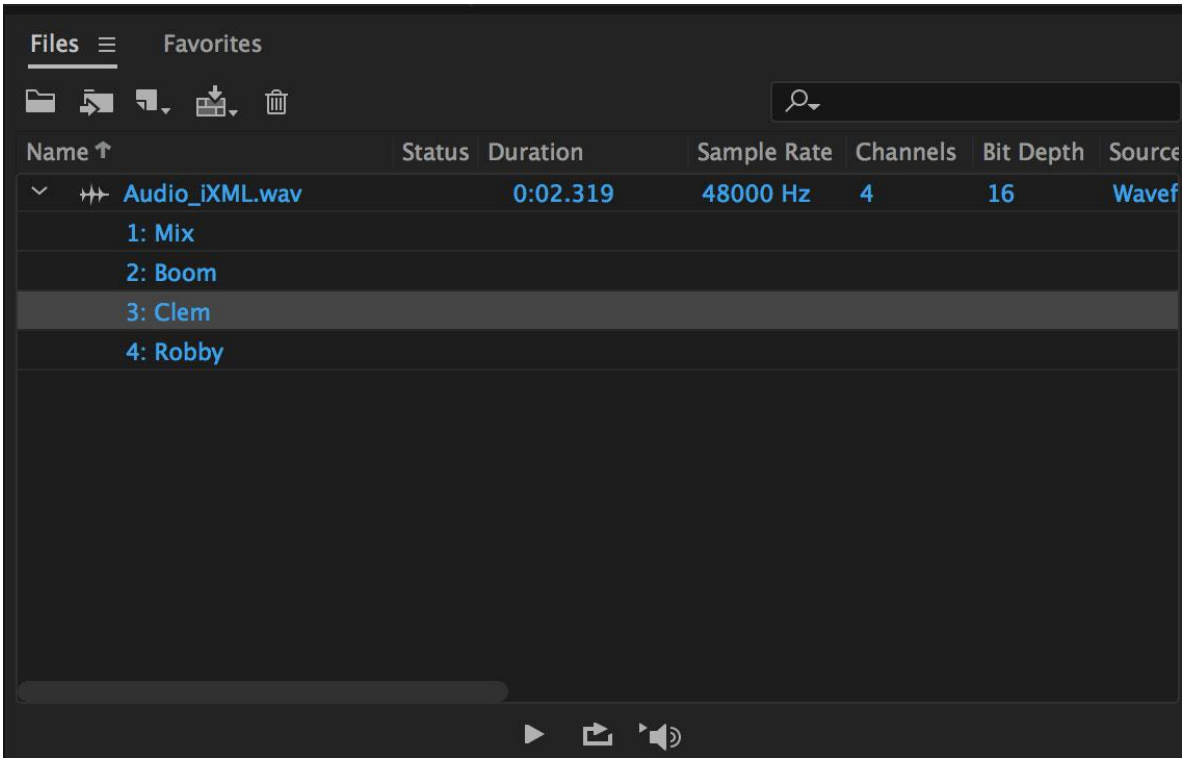


Channel list

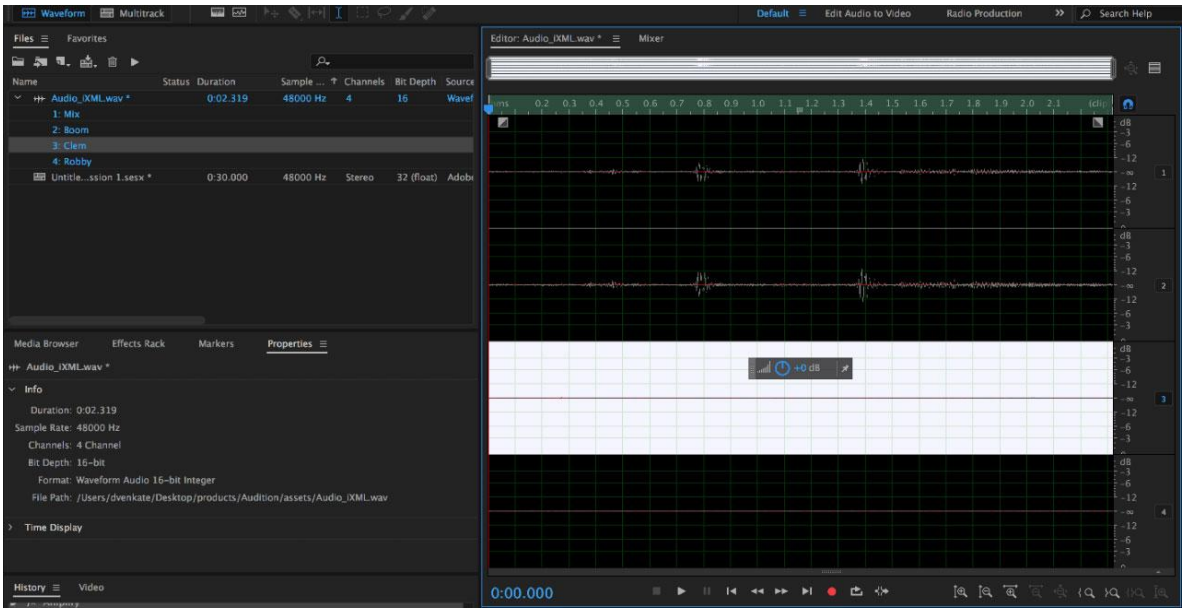
3 To disable other channels, double-click on an individual channel.

Note:

Channel names from iXML metadata would be shown if available.



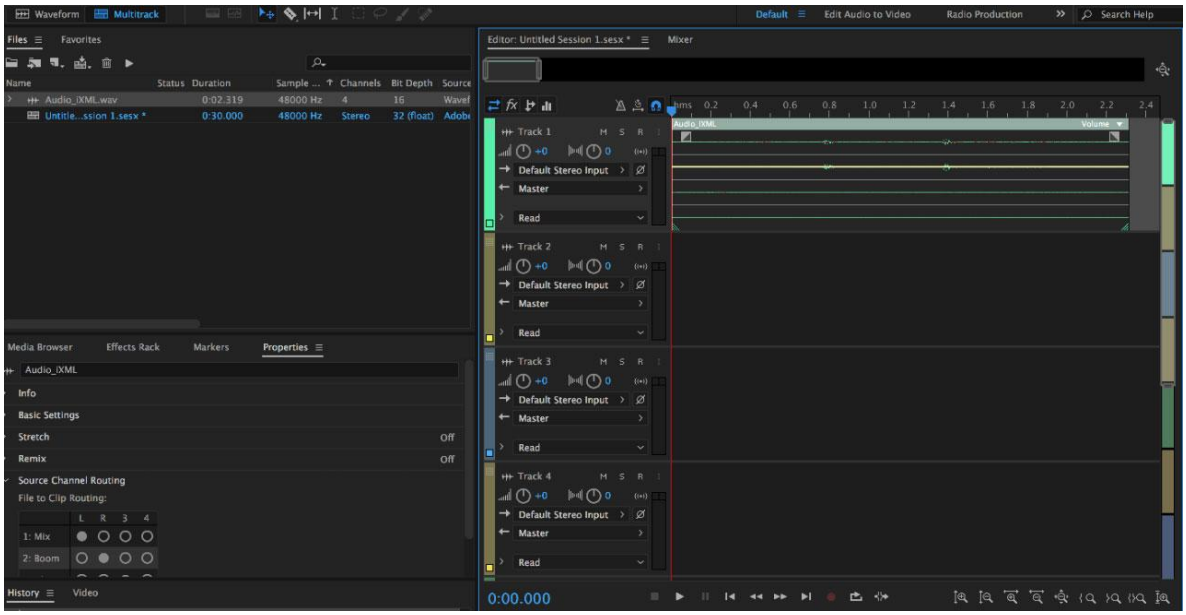
Enable specific channels



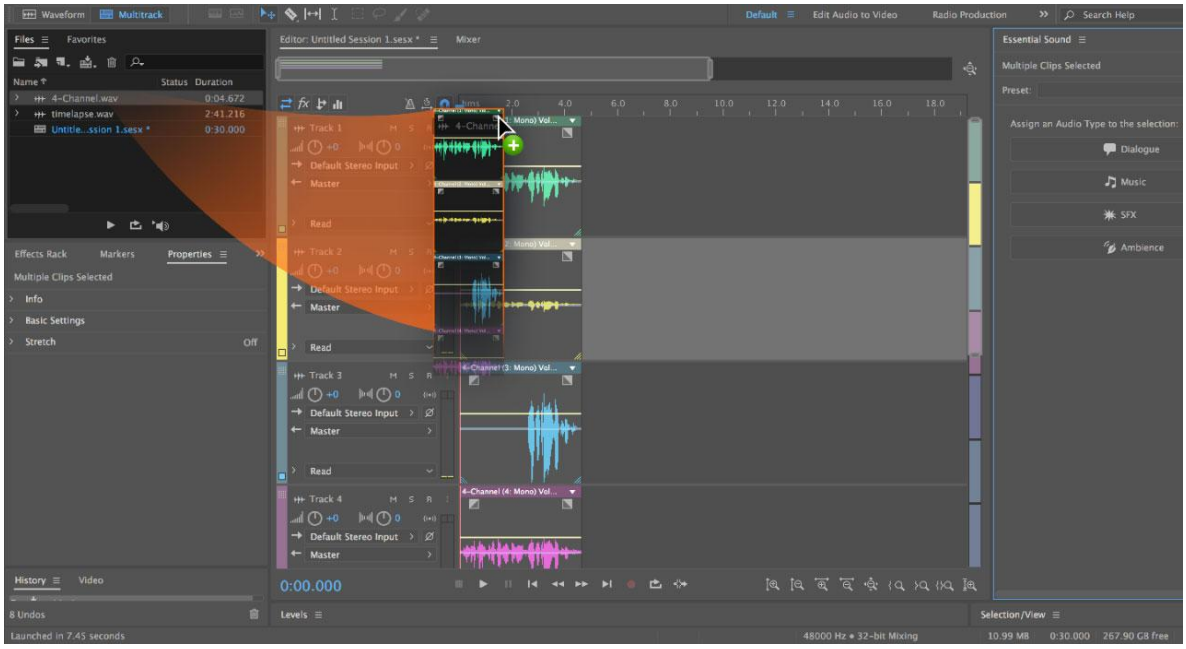
Waveform editor

Multitrack Editor

To use multichannel audio or video files within a session, you can bring each of the channels of the file into the **Multitrack Editor** as one single multichannel clip (default behavior). You can also automatically split each channel or groups of channels into different clips by holding **Alt** (Windows) or **Option** (Mac) while dragging.

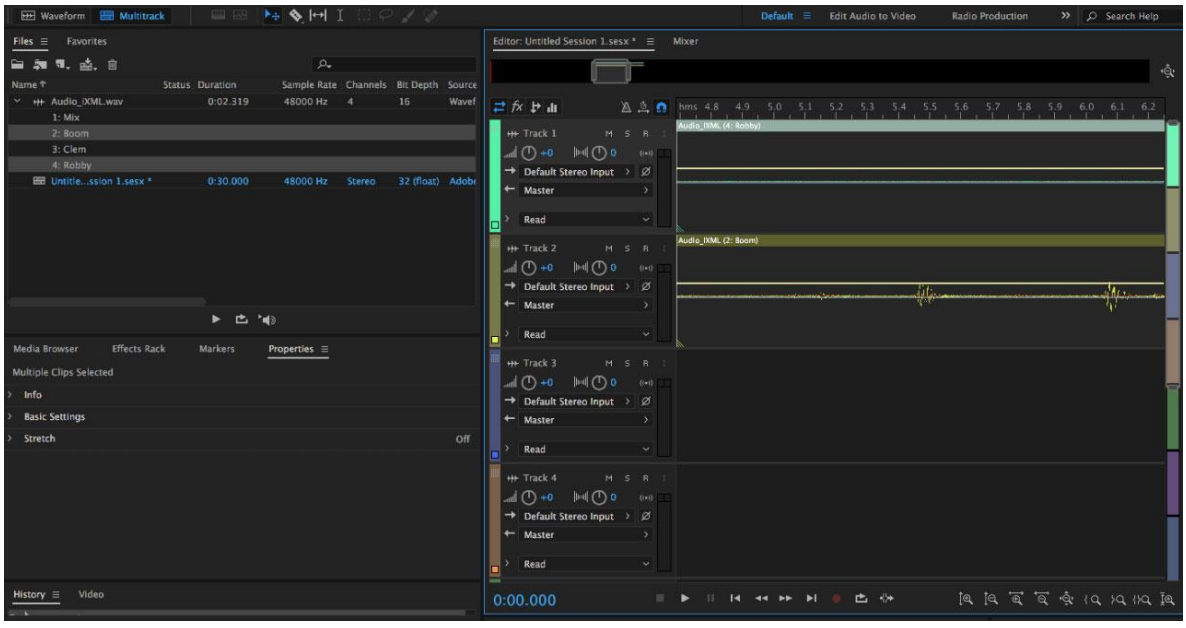


Importing audio as one multichannel clip



Importing audio to Multitrack Editor as different clips

- Select and drag the channels you want to import to the **Multitrack** panel. You can also select multiple channels and drag them to the same track or different tracks.
- To add channels to different tracks, hold **Alt** (Windows) or **Option** (MAC).



Insert clip to different tracks

Note:

*To channelize the clip according to the channelization of the destination track, use the double-modifier behavior using **Command+Option** (MAC) and **Control+Alt** (Windows).*

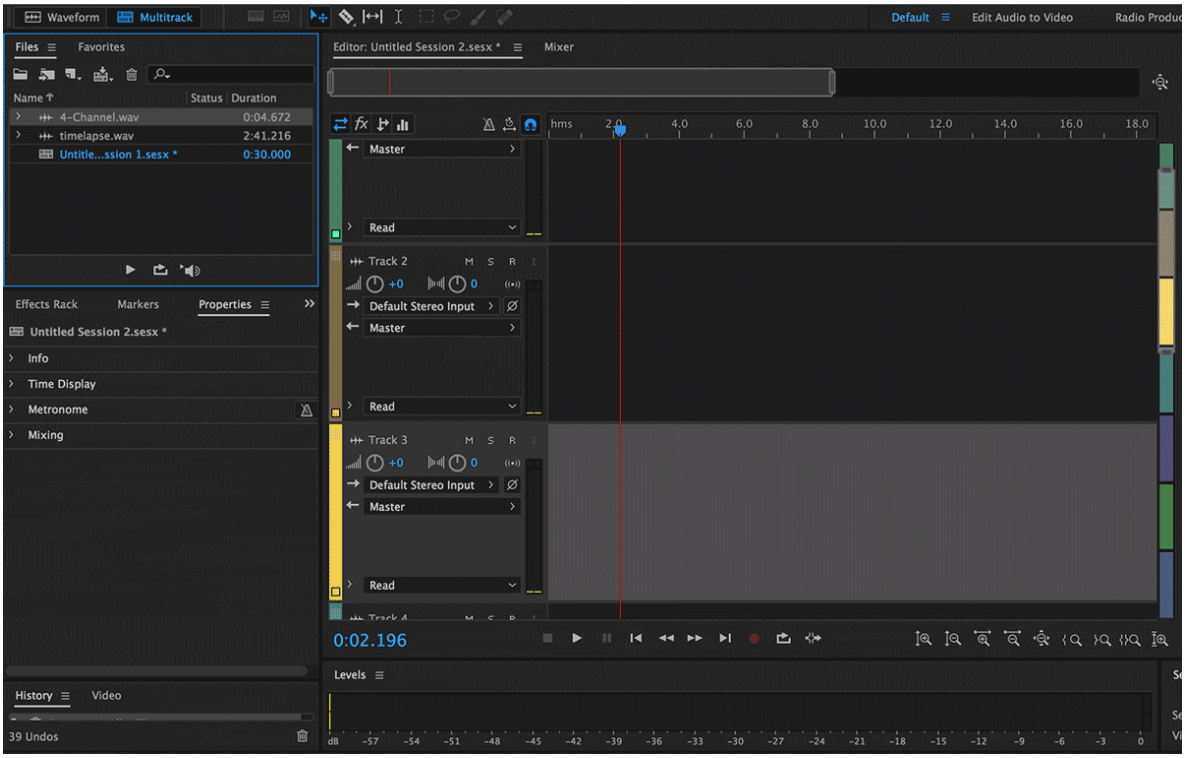
Modify the channels in a specific clip

To modify the channels you have added to specific clip, follow these steps.

Note:

You cannot modify the number of channels in a clip you have added to a track. You can, however, replace one channel with another.

1. Select the clip you want to edit.
2. In the **Properties** panel, select **Source Channel Routing.**
3. Use radio buttons to modify the source channel routing of the selected clip

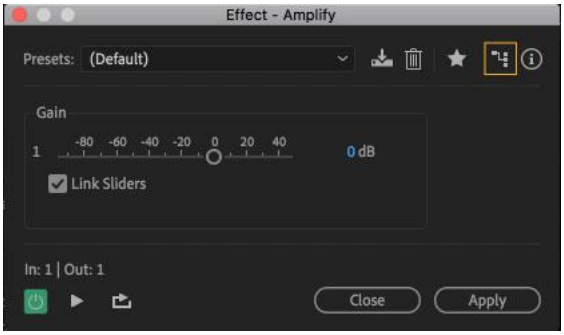


Specific Channel Routing

Effects channelization

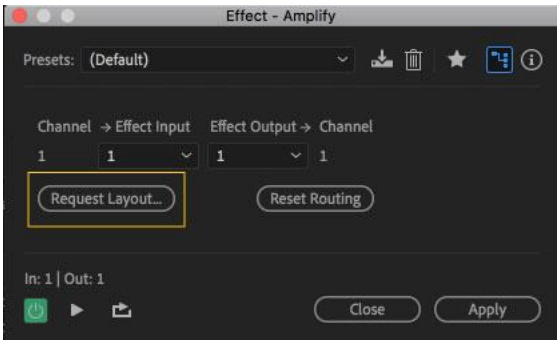
While working with effects, you can assign different in/out channels, and query for the different channel configurations they support. Easily route specific audio clip or track channels to mono, stereo, or surround/immersive effects, and quickly see at-a-glance existing routing configurations.

1. To modify effect layout, open an effect from the Effects Rack or Effects menu. The Effects window opens. Now, select the Channel map editor.



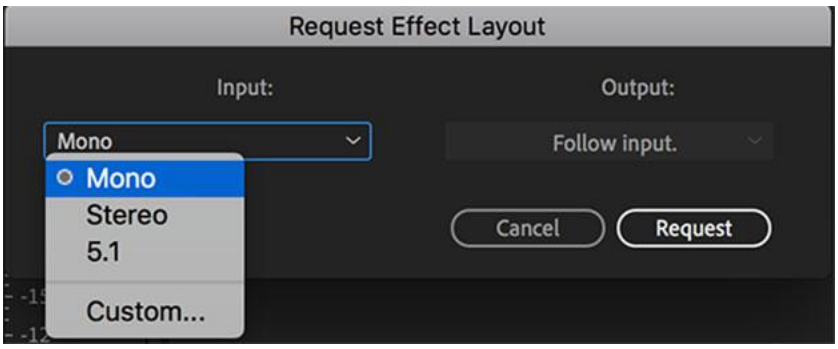
Channel map editor

2 Choose Request Layout. The Request Effect Layout window opens.



Request layout

3 The new Request Effect layout window allows you to choose the input type. Choose from Mono, Stereo, and 5.1 from the drop-down.

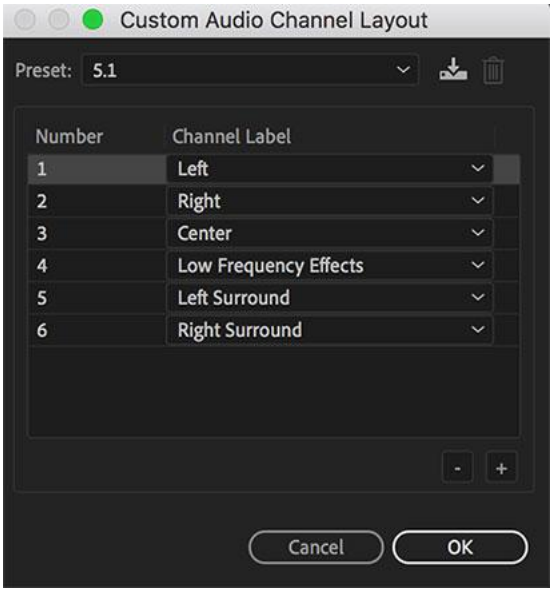


Choose input from drop-drown

Note:

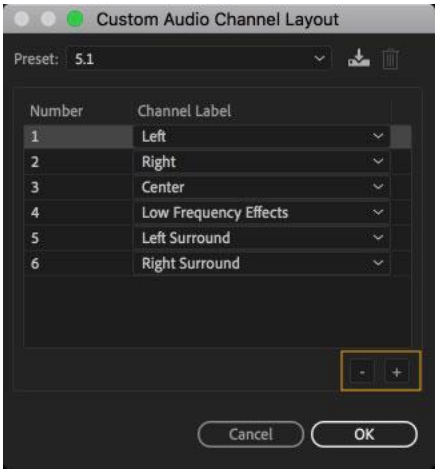
The output is always follows the input and cannot be changed.

4 For custom channel configurations, choose Custom and assign or edit Channel Label



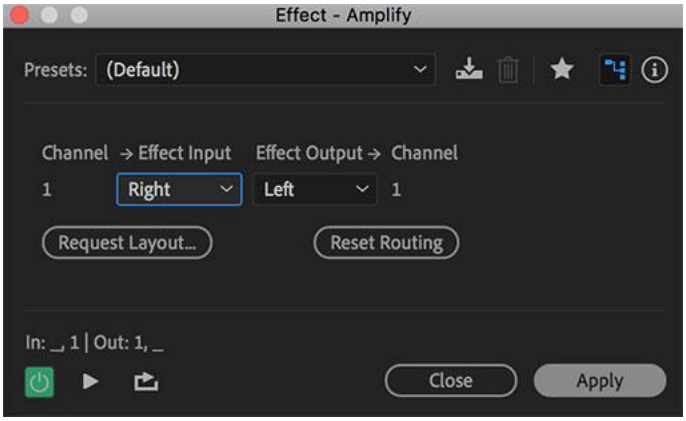
Custom audio channel layout

5 Use the plus and minus icons shown below to add and remove channels.



Add or remove channels

6 You can now modify the effect input and effect output for all individual channels.



Modify effect input and effect output

Create, open, or import files in Adobe Audition

How to Create, Open, and Import files to Adobe Audition

How to Create, Open, and Import files to Adobe Audition

Create a new blank audio file

A new blank audio file is perfect for recording new audio or combining pasted audio.

1 Choose File > New > Audio File.

To quickly create a file from selected audio in an open file, choose Edit > Copy To New. (See Copy or cut audio data.)

2 Enter a filename, and set the following options:

Sample Rate Determines the frequency range of the file. To reproduce a given frequency, the sample rate must be at least twice that frequency. (See Understanding sample rate.)

Channels Determines if the waveform is mono, stereo, 5.1 surround. Adobe Audition saves the last five custom audio channel layouts that you had used for quick access.

For voice-only recordings, the mono option is a good choice that results in quicker processing and smaller files.

Bit Depth Determines the amplitude range of the file. The 32-bit level provides maximum processing flexibility in Adobe Audition. For compatibility with common applications, however, convert to a lower bit depth when editing is complete. (See Understanding bit depthand Change the bit depth of a file.)

Create a multitrack session

Session (*.sesx) files contain no audio data themselves. Instead, they are small XML-based files that point to other audio and video files on the hard drive. A session file tracks the following:

- Files which are a part of the session.
- Location of these files.
- Envelopes and effects that are applied.

To examine settings in detail, .sesx files can be opened in text editors or stored in version control systems. You can use Perforce or Git, which are popular in the gaming industry.

1. Choose File > New > Multitrack Session.
2. Enter a filename and location, and set the following options:

   Template Specifies either a default template or one you've created. Session templates specify source files and settings such as Sample Rate and Bit Depth.

   Sample Rate Determines the frequency range of the session. To reproduce a given frequency, the sample rate must be at least twice that frequency. (See Understanding sample rate.)

   Note:

   All files added to a session must share the sample rate. If you attempt to import files with different sample rates, Adobe Audition prompts you to resample them, which can reduce audio quality. To change resampling quality, adjust the Sample Rate Conversion settings in the Data preferences.

   Bit Depth Determines the amplitude range of the session, including recordings and files created with the Multitrack > Mixdown To New File command. (See Understanding bit depth.)

   Note:

   Choose a bit depth carefully, because it cannot be changed after you create a session. Ideally, work at the 32-bit level with fast systems. If your system performs slowly, try a lower bit depth.

   Master Determines whether tracks are mixed down to a mono, stereo, or 5.1 Master track. (See Routing audio to buses, sends, and the Master track.)

Open existing audio files and multitrack mixes

The following file types open in the Multitrack Editor: Adobe Audition Session, Adobe Audition 3.0 XML, Adobe Premiere Pro Sequence XML, Final Cut Pro XML Interchange, and OMF.

All other supported file types open in the Waveform Editor, including the audio portion of video files.

Note:

SES session files from Adobe Audition 3.0 and earlier are unsupported. If you have Adobe Audition 3.0, save sessions to XML format to open them in later versions.

If you open multiple files, Editor panel menu lets you choose which file to display

1 Choose File > Open.

2 Select an audio or video file. (See Supported import formats.)

If you don't see the file that you want, choose All Supported Media from the menu at the bottom of the dialog box.

Append audio files to another

Append files with CD Track markers to quickly assemble audio and apply consistent processing.

1 In the Waveform Editor, do either of the following:

- To add to the active file, choose File > Open Append > To Current.
- To add to a new file, choose File > Open Append > To New.

2

In the Open Append dialog box, select one or more files.

Note:

If the files have different sample rates, bit depth, or channel type, Adobe Audition converts the selected files to match the open file. For the best results, append files with the same sample type as the original file.

3 Click Open.

Import a file as raw data

You cannot open files which do not have header information that describes the sample type. To manually specify this information, import the file as raw data.

1 Choose File > Import > Raw Data.

2 Select the file, and click Open.

3 Set the following options:

Sample Rate Match the known rate of the file, if possible. For examples of common settings, see Understanding sample rate. Adobe Audition can import raw data with rates ranging from 1 Hz to 10,000,000 Hz. But playback and recording are supported only between 6000 Hz and 192,000 Hz.

Channels Enter a number from 1 to 32.

Encoding Specifies the data storage scheme for the file. If you are unsure what encoding the file uses, consult the supplier of the file, or the documentation for the application that created it.

Byte Order Specifies the numerical sequence for bytes of data. The Little-Endian method is common to WAV files, while the Big-Endian method is common to AIFF files. The Default Byte Order automatically applies the default for your system processor and is typically the best option.

Start Byte Offset Specifies the data point in the file at which the import process begins.

Insert an audio file into a multitrack session

When you insert an audio file in the Multitrack Editor, the file becomes an audio clip on the selected track. If more than one file is inserted, or if the file is longer than the space available on the selected track, a new clip is inserted. The clip is inserted to the nearest empty track.

1 In the Multitrack Editor, select a track, and then place the playhead at the desired time position.

2 Choose Multitrack > Insert Files.

3 Select an audio or video file. (See Supported import formats.)

Drag ranges from the Markers panel to the Multitrack Editor to automatically convert them to clips.

Spot-insert a Broadcast Wave file into a session

When you insert a Broadcast Wave (BWF) file into a multitrack session, Adobe Audition can use the embedded timestamp to insert the file at a specific time. The action is commonly called spot-inserting.

1 Choose Edit > Preferences > Multitrack (Windows) or Audition > Preferences > Multitrack (Mac OS).

2 Select Use Embedded Timecode When Inserting Clips Into Multitrack.

3 In the Multitrack Editor, select a track.

4 Choose Multitrack > Insert Files, and select one or more BWF files.

Adobe Audition inserts an audio clip at the designated start time.

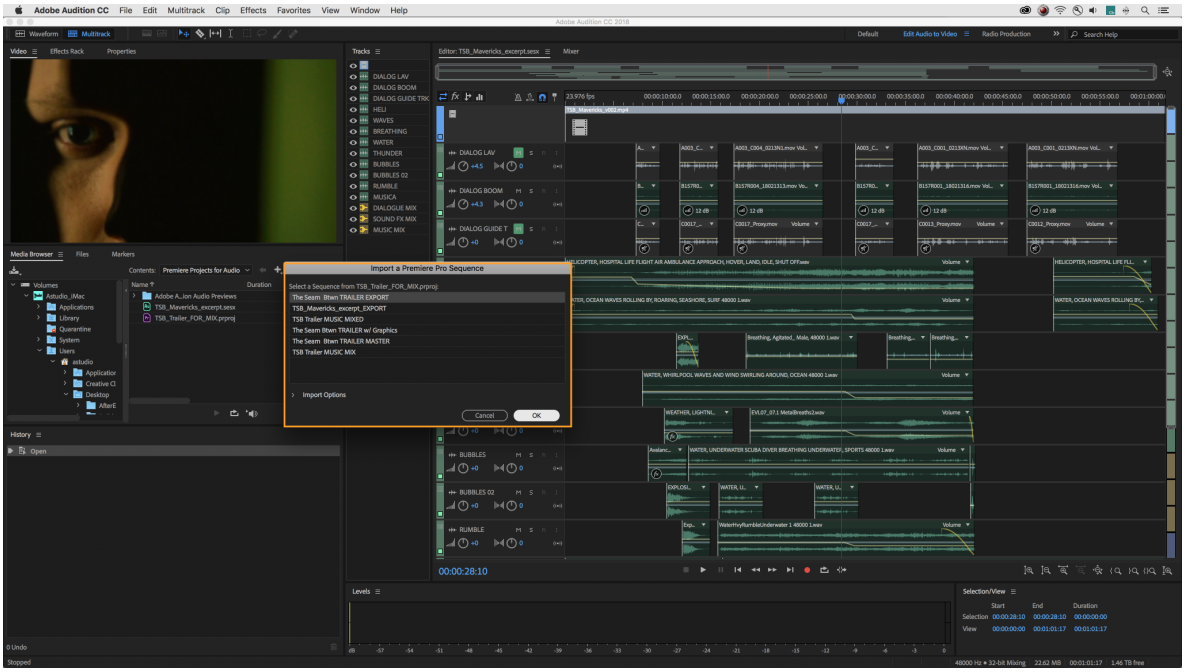
To view or edit the timestamp for a BWF clip, open the clip in the Waveform Editor, and then choose Window > Metadata. On the BWF tab, the timestamp value appears as the Time Reference.

Import sequence from Adobe Premiere Pro

You can directly import Adobe Premiere Pro projects (.prproj) to Adobe Audition. This import method uses the original media and does not require rendering.

1 Click **File > Import > File** and choose the Adobe Premiere Pro project to import.

2 The **Import a Premiere Pro Sequence** dialogue box opens with the list of sequences the project contains. You can select the specific sequence that you want to open. The selected sequence is directly imported referencing the original media.

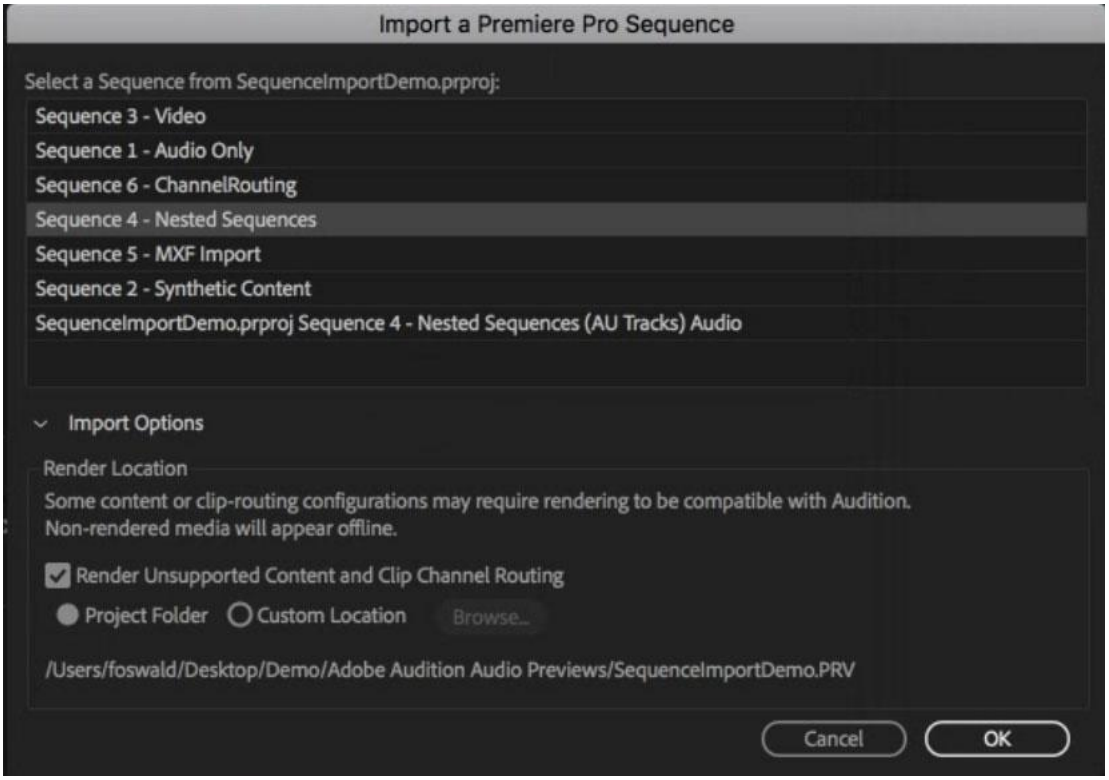


Import Adobe Premiere Pro Sequence

3 Some content or clip-routing configuration requires the audio to be rendered. For example, synthetic content, nested sequences, and incompatiblechannel-routing configuration require rendering. If you choose to not render these sequences, they appear offline.

4 To enable or disable the **Render Unsupported Content and Clip Channel Routing** check box, do the following:

- Open the **Import Options** drop-down on the **Import a Premiere Pro sequence** dialogue box.


Import a Premiere Pro Sequence
Select a Sequence from SequenceImportDemo.prproj:
Sequence 3 - Video
Sequence 1 - Audio Only
Sequence 6 - ChannelRouting
Sequence 4 - Nested Sequences
Sequence 5 - MXF Import
Sequence 2 - Synthetic Content
SequenceImportDemo.prproj Sequence 4 - Nested Sequences (AU Tracks) Audio
Import Options
Render Location
Some content or clip-routing configurations may require rendering to be compatible with Audition.
Non-rendered media will appear offline.
Render Unsupported Content and Clip Channel Routing
Project Folder
Custom Location
Browse...
/Users/foswald/Desktop/Demo/Adobe Audition Audio Previews/SequenceImportDemo.PRV
Cancel
OK

Import options

5 You can select the location to save the rendered content using the **Project Folder** and **Custom Location** radio buttons. Click **Browse** to select a custom location.

The sequences are imported to Adobe Audition with reference to original media. The imported video from Adobe Premiere Pro is shown as a single flattened layer.

Importing with the Files panel

The Files panel displays a list of open audio and video files for easy access.

Double-click an empty area of the file list to quickly access the Open File dialog box.

Import files into the Files panel

Import files into the Files panel if you want to retain the currently open file in the Editor panel. This technique is particularly helpful when assembling files for a multitrack session.

1 Do either of the following:

- In the Files panel, click the Import File button .
- Choose File > Import > File.

2 Select an audio or video file. (See Supported import formats.)

Insert from the Files panel into a multitrack session

1 In the Files panel, select the files you want to insert.

To select multiple adjacent files, click the first file in the desired range, and then Shift-click the last. To select nonadjacent files, Ctrl-click (Windows) or Command-click (Mac OS).

2 At the top of the Files panel, click the Insert Into Multitrack button . Then choose either New Multitrack Session (see Create a new multitrack session) or an open session.

The files are inserted on separate tracks at the current time position.

Change displayed metadata in the Files panel

1 In the upper right of the Files panel, click the menu icon , and choose Metadata Display.

2 Select the metadata you want to display, and click OK

3 To move metadata columns left or right, drag column headers such as Name or Duration.

To change the sort order of files, click column headers.

Extracting audio from CDs

Extract CD tracks with the Extract Audio From CD command

The Extract Audio From CD command is faster and provides more control, including the ability to optimize drive speed and rename tracks.

1 Place an audio CD in the computer's CD-ROM drive.

2 Choose File > Extract Audio From CD.

3 For Drive, choose the drive that contains the audio CD.

4 For Speed, choose from all the extraction speeds that the selected drive supports. The Maximum Speed option usually produces satisfactory results, but if it produces errors, specify a slower speed.

5 Do any of the following:

- To preview a track, click its Play button.
- To include or exclude tracks, click the checkboxes to the left of track numbers, or click Toggle All.
- To rename a track, double-click it.

6 Audition CC only: Enable Extract to Single File to create a single file that contains all selected tracks.

Configure track information and CD databases

In the center of the Extract Audio From CD dialog box, Artist, Album, Genre, and Year information is drawn from the specified CD database by default. To adjust these entries, do any of the following:

- To customize information, overwrite current entries.
- To insert original information from the database, click the Retrieve Titles icon .
- If a message indicates multiple matching records, click the arrows to insert different database records.
- To specify a different database and file naming convention, click the Title Settings icon . For details about each Title Settings option, position the mouse over it until a tool tip appears.

Note: If multiple Artist entries are detected, Audition automatically selects the Compilation option. In the Title Settings dialog box, enter a Separator For Compilations character to separate Artist and Track Title for each track.

Extract CD tracks with the Open command (Mac OS)

Note:

This extraction method can greatly degrade Audition performance. The Extract Audio From CD command is usually a better choice.

The Open command lets you extract tracks in AIFF format but requires Audition to continue reading audio data from CD, slowing importing and editing.

1. Place an audio CD in the computer's CD-ROM drive.
2. Choose File > Open.
3. Choose QuickTime as the file type, and navigate to the CD-ROM drive.
4. Select the tracks you want to extract, and click Open.

Supported import formats

Audio import formats

Adobe Audition can open audio files in the following formats:

- AAC (including HE-AAC)
- AIF, AIFF, AIFC (including files with up to 32 channels)

There are many different variations of AIFF format. Audition can open all uncompressed AIFF files and most common compressed versions.

Note:

To see Author metadata in AIFF files, view the Dublin Core: Creator field on the XMP tab of the Metadata panel. (See Viewing and editing XMP metadata.)

- AC-3
- APE
- AU

- AVR
- BWF
- CAF (all uncompressed and most compressed versions)
- EC-3
- FLAC
- HTK
- IFF
- M4A
- MAT
- MPC
- MP2
- MP3 (including MP3-surround files)
- OGA, OGG
- PAF
- PCM
- PVF
- RAW
- RF64
- SD2
- SDS
- SF
- SND
- VOC
- VOX
- W64
- WAV (including files with up to 32 channels)

There are many different variations of WAV format. Adobe Audiition can open all uncompressed WAV files and most common compressed versions.

- WMA (Windows only, and enabled with **DLMS Format Support** in **Media & Disk Cache** preferences)
- WVE
- XI

Video import formats

The Waveform Editor lets you open the audio portion of video files in the formats below. The Multitrack Editor lets you insert the same file types and provides a preview in the Video panel.

To access these video formats, install QuickTime. To import additional formats, extend QuickTime support. For more information, see this article on the Apple website.

- AVI
- DV
- MOV
- MPEG-1
- MPEG-4
- 3GPP and 3GPP2

Note:

*The following formats are enabled with **DLMS Format Support** in **Media & Disk Cache** preferences. This preference is turned on by default.*

- AVI (Windows only)
- FLV
- R3D
- SWF
- WMV

Navigate time and playing audio in Adobe Audition

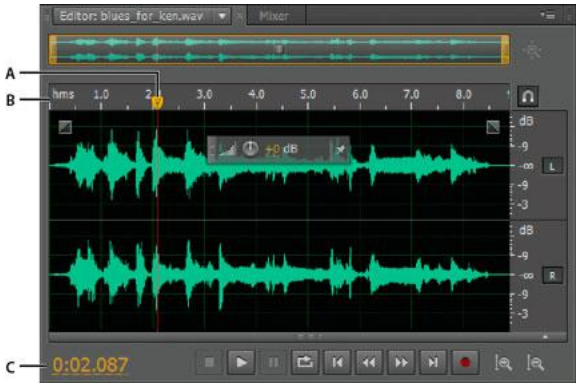
Monitoring time

In the Editor panel, the following features help you monitor time:

1. In the timeline near the top of the panel, the *current-time indicator* lets you start playback or recording at a specific point.
2. In the lower left of the panel, the time display shows the current time in numerical format. The default time format is Decimal, but you can easily change it. (See Change the time display format.) The same format is used by the timeline.

   Note:

   To show the time display in separate panel, choose Window > Time.



Features that help you monitor time
***A** Current-time indicator* ***B** Timeline* ***C** Time display*

Position the current-time indicator

In the Editor panel, do any of the following:

1. In the timeline, drag the indicator or click a specific time point.
2. In the time display at lower left, drag across the numbers, or click to enter a specific time.
3. At the bottom of the panel, click one of the following buttons:

   Note:

   To display these buttons in a separate panel, choose Window > Transport.

   Pause Temporarily stops the current-time indicator. Click the Pause button again to resume playback or recording.

   Move CTI to Previous Places the current-time indicator at the beginning of the next marker. If there are no markers, the current-time indicator moves to the beginning of the waveform or session.

   Rewind Shuttles the current-time indicator backward in time.

   Note:

   Right-click the Rewind button to set the rate at which the cursor moves.

   Fast Forward Shuttles the current-time indicator forward in time.

   Note:

   Right-click the Fast Forward button to set the rate at which the cursor moves.

   Move CTI to Next Moves the current-time indicator to the next marker. If there are no markers, the current-time indicator moves to the end of the waveform or session.

Preview audio by scrubbing

To scrub audio (producing an audible preview as you shuttle across a file), do any of the following:

1. Drag the current-time indicator .
2. Press the Rewind or Fast Forward buttons.
3. Press the J, K, and L keys to shuttle backward, stop, or shuttle forward. Repeatedly pressing the J or L key gradually increases shuttle speed. (To change the default, set JKL Shuttle Speed in the Playback preferences.)

Play audio linearly or in a loop

To quickly start and stop playback, press the spacebar.

1 In the Editor panel, position the current-time indicator, or select a range.

2 (Optional) At the bottom of the panel, right-click the Play button , and select one of the following:

Return CTI to Start Position on Stop Reflects the behavior of Audition 3.0 and earlier. (Press Shift+X to toggle this option on and off.)

Play Spectral Selection Only Plays only frequencies you've selected with the Marquee , Lasso , or Paintbrush Selection tool.

3 (Optional) Click the Loop Playback button if you want to fine-tune a selected range or experiment with different effects processing.

4 To start playback, click the Play button.

Note:

By default, the Editor panel scrolls when playback extends beyond the visible section of a waveform. In the Playback area of the Preferences dialog box, you can disable auto-scrolling.

Synchronize the current-time indicator across files or views

In the Waveform Editor, you can maintain the position of the current-time indicator when you switch between files—a useful technique when editing different versions of the same waveform. In the Multitrack Editor, you can maintain the position of the current-time indicator when you switch to the Waveform Editor—a useful technique when applying edits and effects in both views.

Synchronize the current-time indicator between files in the Waveform Editor

1 Choose Edit > Preferences > General (Windows) or Adobe Audition Preferences > General (Mac OS).

2 Select Synchronize Selection, Zoom Level, and CTI Across Files In The Waveform Editor.

Synchronize the current-time indicator between the Multitrack and Waveform Editors

1 Choose Edit > Preferences > Multitrack (Windows) or Adobe Audition Preferences > Multitrack (Mac OS).

2 Select Synchronize Clips With Waveform Editor.

Change the time display format

By default, all audio files and multitrack sessions use the same time display format. To customize the format for an open file or session, choose Window > Properties, expand the Advanced settings, and deselect Synchronize With Time Display Preferences.

Choose View > Display Time Format, and choose the desired option:

Decimal (mm:ss.ddd) Displays time in minutes, seconds, and thousandths of a second.

Compact Disc 75 fps Displays time in the same format used by audio compact discs, where each second equals 75 frames.

SMPTE 30 fps Displays time in the SMPTE format, where each second equals 30 frames.

SMPTE Drop (29.97 fps) Displays time in the SMPTE drop-frame format, where each second equals 29.97 frames.

SMPTE 29.97 fps Displays time in the SMPTE non-drop-frame format, where each second equals 29.97 frames.

SMPTE 25 fps (EBU) Displays time using the European PAL television frame rate, where each second equals 25 frames.

SMPTE 24 fps (Film) Displays time in a format where each second equals 24 frames, suitable for film.

Samples Displays time numerically, using as a reference the actual number of samples that have passed since the beginning of the edited file.

Bars and Beats Displays time in a musical measures format of bars:beats: subdivisions. To customize settings, choose Edit Tempo, and set the following options in the Properties panel.

Tempo Specifies beats per minute.

Custom (X frames per second) Displays time in a custom format. To modify a custom format, choose Edit Custom Frame Rate, and enter a number of frames per second. Valid values are whole numbers from 2 to 1000.

Jump to time

Use the Jump to time shortcut to set the focus to the timecode display. This allows you to enter a timecode to position the play-head. To enter timecode, follow these steps:

1. In the time display at lower left, drag across the numbers, or click to enter a specific time.
2. The timecode is highlighted. Enter the time to jump the play-head to.
3. Press Enter. The timecode moved to the specified time.

Recording audio

Record audio in the Waveform Editor

You can record audio from a microphone or any device you can plug into the Line In port of a sound card. Before recording, you have to adjust the input signal to optimize signal-to-noise levels. (See either Adjust recording levels for standard sound cardsor the documentation for a professional card.)

1. Set audio inputs. (See Configure audio inputs and outputs.)
2. Do one of the following:
   - Create a file.
   - Open an existing file to overwrite or add new audio, and place the current-time indicator where you want to start recording.
3. At the bottom of the Editor panel, click the Record button to start and stop recording.

Correct DC offset

Some sound cards record with a slight *DC offset*, in which direct current is introduced into the signal, causing the center of the waveform to be offset from the zero point (the center line in the waveform display). DC offset can cause a click or pop at the beginning and end of a file.

? In the Waveform Editor, choose Favorites > Repair DC Offset.

To measure DC offset, see Analyze amplitude.

Direct-to-file recording in the Multitrack Editor

In the Multitrack Editor, Adobe Audition automatically saves each recorded clip directly to a WAV file. Direct-to-file recording lets you quickly record and save multiple clips, providing tremendous flexibility.

Inside the session folder, you find each recorded clip in the [session name]_Recorded folder. Clip filenames begin with the track name, followed by the take number (for example, Track 1_003.wav).

After recording, you can edit takes to produce a polished final mix. For example, if you create multiple takes of a guitar solo, you can combine the best sections of each solo. (See Trimming and extending clips.) Or, you can use one version of the solo for a video soundtrack, and another version for an audio CD.

Record audio clips in the Multitrack Editor

In the Multitrack Editor, you can record audio on multiple tracks by *overdubbing*. When you overdub tracks, you listen to previously recorded tracks and play along with them to create sophisticated, layered compositions. Each recording becomes a new audio clip on a track.

1. In the Inputs/Outputs area of the Editor panel, choose a source from a track's Input menu.

   Note:

   To change the available inputs, choose Audio Hardware, and then click Settings.

2. Click the Arm For Record button for the track.

   The track meters display the input, helping you optimize levels. (To disable this default and display levels only while recording, deselect Enable Input Metering When Arming Tracks in the Multitrack preferences.)

3. To hear hardware inputs routed through any track effects and sends, click the Monitor Input button .

   Note:

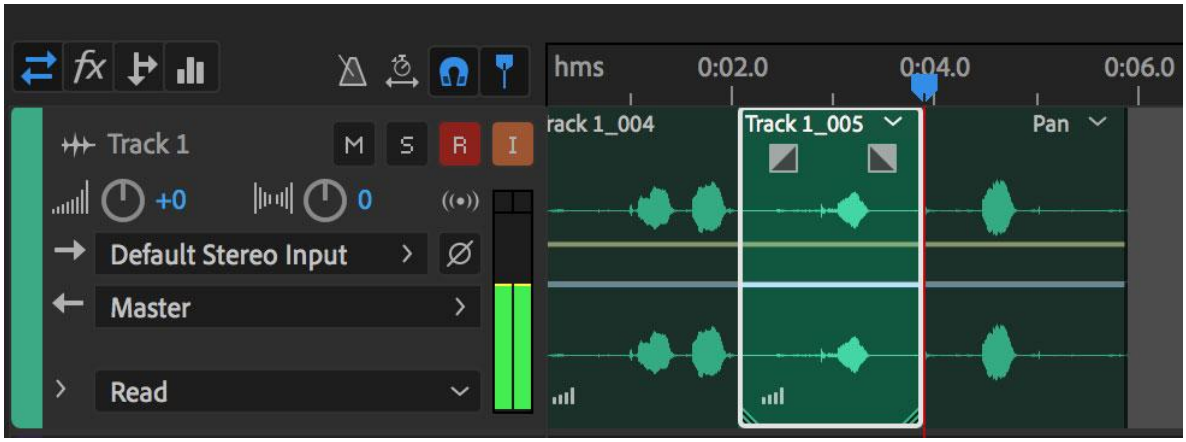
   Routing inputs through effects and sends requires significant processing. To reduce latency (an audible delay) that disrupts timing for performers, see Configure audio inputs and outputs.

4. To simultaneously record on multiple tracks, repeat steps 1-3.
5. In the Editor panel, position the current-time indicator at the desired starting point, or select a range for the new clip.
6. At the bottom of the panel, click the Record button to start and stop recording.

Punch into a selected range

If you're dissatisfied with a time range of a recorded clip, you can select that range and punch in a new recording, leaving the original clip intact. Though you can record into a specific range without punching in, punching in lets you hear audio immediately before and after a range; that audio provides vital context that helps you create natural transitions.

For important or difficult sections, you can punch in multiple takes, and then select or edit takes to create the best performance.



A take created by punching in

1. In the Editor panel, drag the Time Selection tool in the appropriate track to select a time range for the clip.
2. Select the correct track input. (See Assign audio inputs and outputs to tracks.)
3. Click the Arm For Record button for the track.
4. Position the current-time indicator a few seconds before the selected range.
5. At the bottom of the Editor panel, click the Record button.

Adobe Audition plays the audio preceding the selection, records during the selected range, and then resume playback.

Punch in during playback

If you don't wish to punch a specific range, you can quickly punch into a general area during playback.

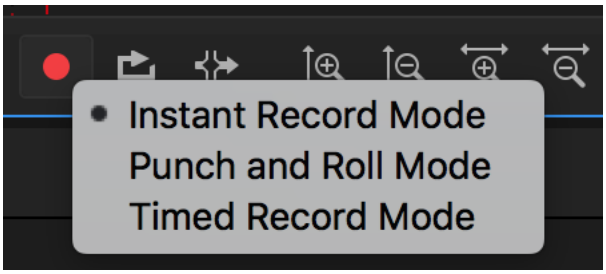
1. Enable one or more tracks for recording. (See Record audio clips in the Multitrack Editor.)
2. At the bottom of the Editor panel, click the Play button.
3. When you reach an area where you want to begin recording, click the Record button. When you finish recording, click the button again.

Punch and Roll

This record mode allows you to record audio using the punch and roll technique with pre-roll, visual countdown, and Punch Again functionality. When recording is done on this mode, playback runs for the specified pre-roll duration before the recording punches in at the desired point. The pre-roll duration can be configured on the Playback and Recording Preferences page.

To record, follow these steps:

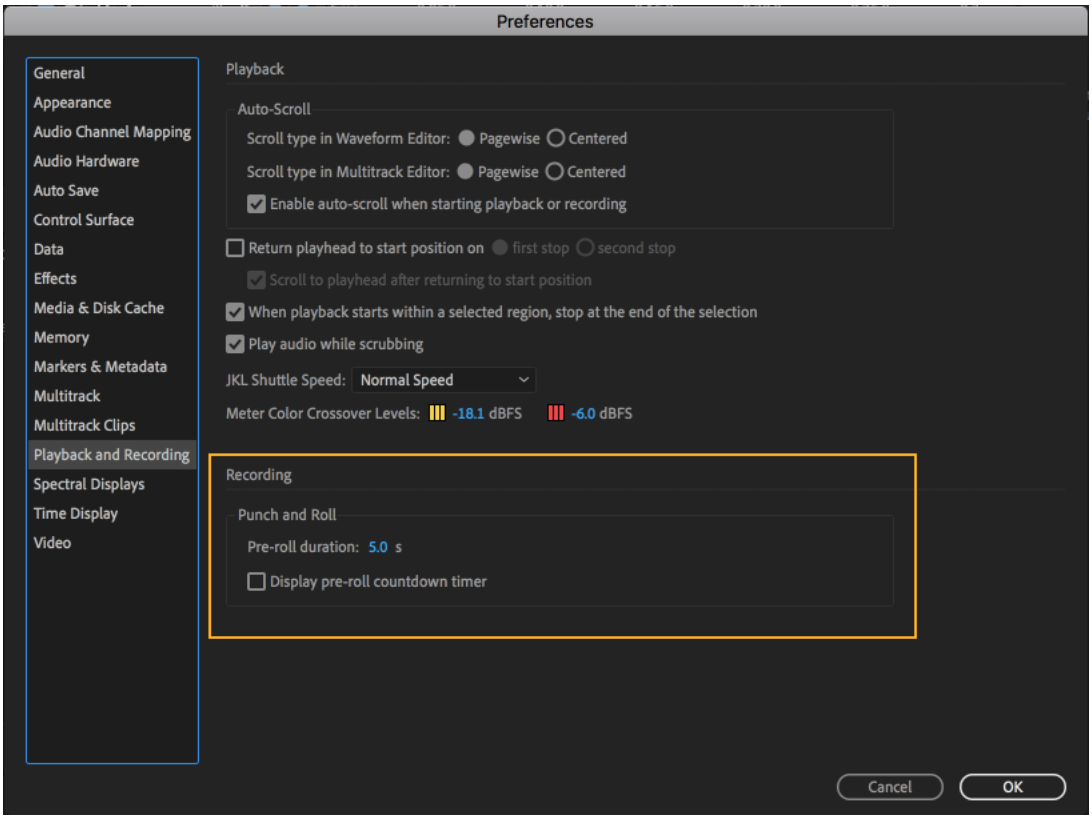
1 Toggle Punch and Roll recording mode by right-clicking the Record button and selecting **Punch and RollMode**. You can also use the keyboard shortcut Shift+Alt+Space.



Recording modes

2 Position the playhead where you would like to punch in. To begin recording, press Record.

3 To adjust pre-roll duration, open **Preferences > Playback and Recording.** The pre-roll is set to 5 seconds by default.



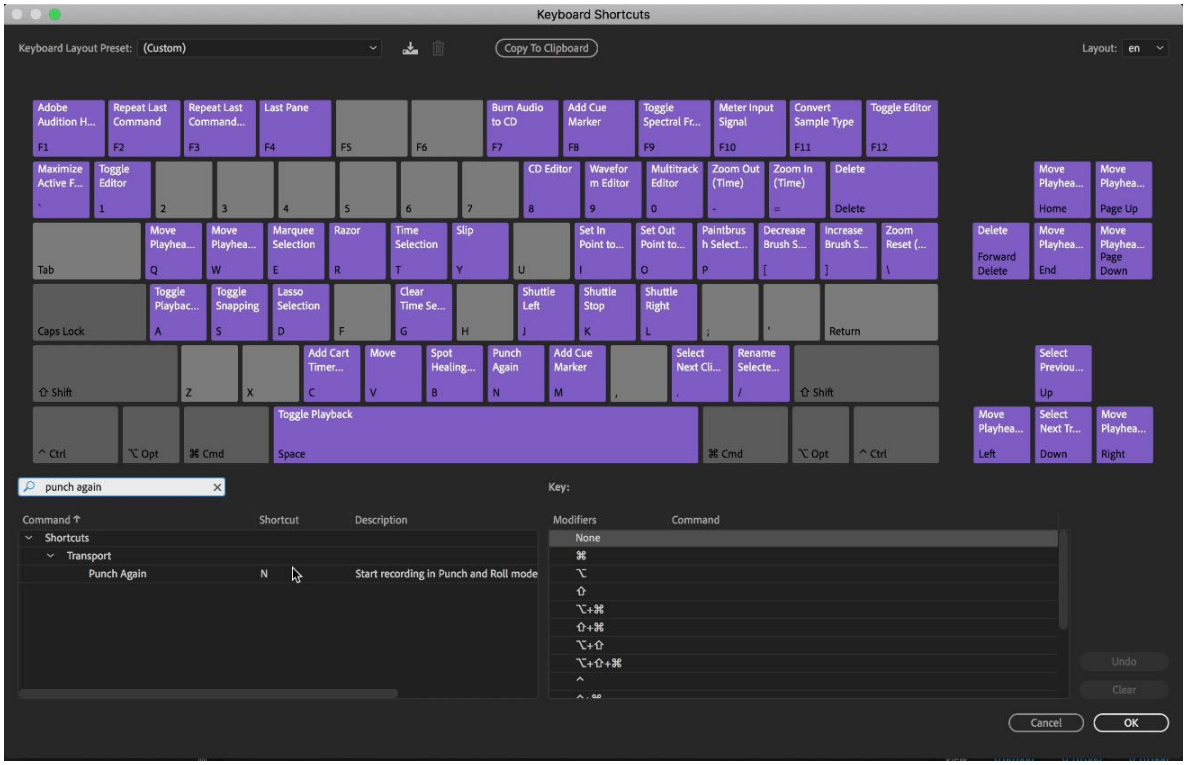
Adjust pre-roll duration

The Punch Again command stops active playback or recording and starts a Punch and Roll recording from the last punch in point. You can configure this command using keyboard shortcuts.

4 For video files, the countdown is displayed in the Video panel as an overlay, with a flash frame when recording begins.

Punch again

Punch Again command stops active playback or recording and starts a Punch and Roll recording from the last punch in point.



Punch again keyboard shortcut

Choose punch-in takes

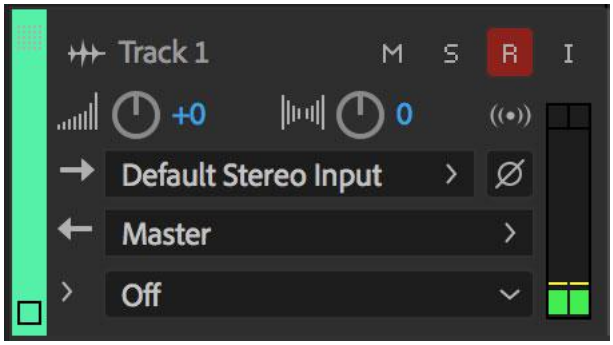
If you punch in multiple takes, Adobe Audition layers the takes over each other in the Editor panel. To choose between takes, do the following:

1. With the Time Selection tool , select a range that snaps to the start and end of the punch-in takes. (See Snap to clip endpoints.)
2. In the track, position the mouse over the clip header. (The header displays the track name, followed by take number.)
3. Drag the topmost take to a different location (typically the end of the session to avoid unwanted playback).
4. Play the session. If you prefer a take you previously moved, drag it back to the selected range.

To mute the original clip for the duration of the punch-in range, adjust the volume envelope. (See Automating clip settings.)

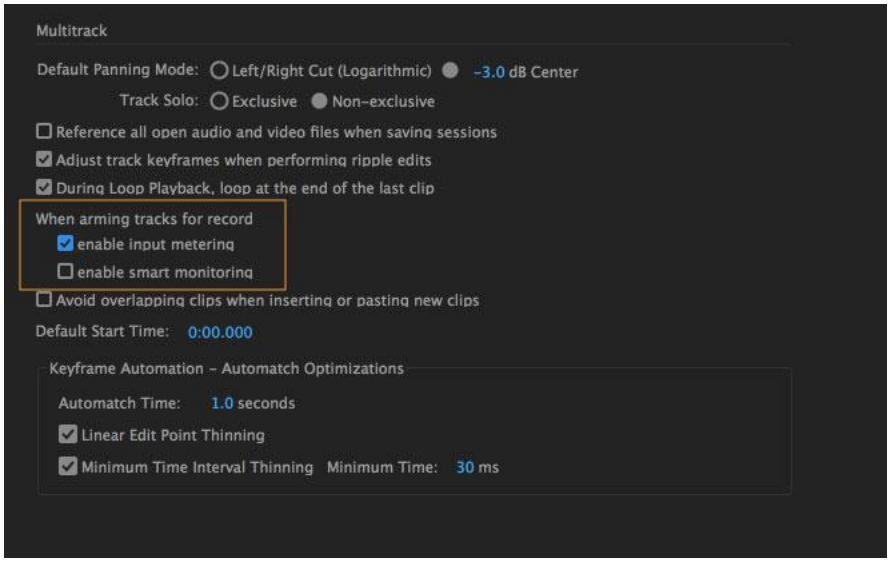
Input monitoring

Input monitoring allows you to preview the incoming input from your recording device. This needs to be done before starting the actual recording by either observing the level meters or listening to it. Click the **[I]** button on the track header controls or in the mixer to enable input monitoring. The **[R]** button arms the track for recording. This affects the audio routing depending on your preferences.



Arm track for record

The track level meters reflect the audio coming from your input device and is routed to the track output. Adobe Audition offers two input monitoring modes that are activated when you arm a track for record.


Multitrack
Default Panning Mode: Left/Right Cut (Logarithmic) -3.0 dB Center
Track Solo: Exclusive Non-exclusive
Reference all open audio and video files when saving sessions
Adjust track keyframes when performing ripple edits
During Loop Playback, loop at the end of the last clip
When arming tracks for record
enable input metering
enable smart monitoring
Avoid overlapping clips when inserting or pasting new clips
Default Start Time: 0:00.000
Keyframe Automation – Automatch Optimizations
Automatch Time: 1.0 seconds
Linear Edit Point Thinning
Minimum Time Interval Thinning Minimum Time: 30 ms

Input metering

• **Input metering:** Enable the input metering check box to view the input levels on the track level meter when the transport is stopped or recording. To enable input metering, follow these steps:

1. Select **Adobe Audition > Preferences > Multitrack**.

2. Enable the enable input metering check box.

For more information about level meters, see Level meters overview.

• **Smart monitoring:** Smart monitoring automatically enables input monitoring during recording, and while the player is stopped, allowing for better overdubs.

While the transport is stopped, you can hear the audio input. For example, communicate with your speaker. After starting playback, the input is disregarded and you only hear the content of the track playing. This helps you find the perfect punch-in point. You can then listen to the recorded audio.

To enable input monitoring, follow these steps:

1. Select **Adobe Audition > Preferences > Multitrack**.

2. Enable the **enable smart monitoring** check box.

Monitoring recording and playback levels

Level meters overview

To monitor the amplitude of incoming and outgoing signals during recording and playback, you use level meters. The Waveform Editor provides these meters only in the Levels panel. The Multitrack Editor provides them in both the Levels panel, which shows the amplitude of the Master output, and track meters, which show the amplitude of individual tracks.

You can dock the Levels panel horizontally or vertically. When the panel is docked horizontally, the upper meter represents the left channel, and the lower meter represents the right channel.

To show or hide the panel, choose Window > Level Meters.

Levels panel, docked horizontally
***A** Left channel **B** Right channel **C** Peak indicators **D** Clip indicators*

The meters show signal levels in *dBFS* (decibels below full scale), where a level of 0 dB is the maximum amplitude possible before clipping occurs. Yellow peak indicators remain for 1.5 seconds so you can easily determine peak amplitude.

If amplitude is too low, sound quality is reduced; if amplitude is too high, clipping occurs and produces distortion. The red clip-indicator to the right of the meters lights up when levels exceed the maximum of 0 dB.

To clear clip indicators, either click them individually, or right-click the meters and choose Reset Indicators.

Customize level meters

Right-click the meters and select any of the following options:

Meter Input Signal In the Waveform Editor, displays the level of the default hardware input. (See Configure audio inputs and outputs.) To quickly enable or disable this option, double-click the meters.

Range options Change the displayed decibel range.

Show Valleys Shows valley indicators at low-amplitude points.

***Tip**: If valley indicators are close to peak indicators, dynamic range (the difference between the quietest and loudest sounds) is low. If the indicators are spread far apart, dynamic range is high.*

Show Color Gradient Gradually transitions the meters from green, to yellow, to red. Deselect this option to display abrupt color shifts to yellow at -18 dBFS, and red at -6.

Show LED Meters Displays a separate bar for each whole decibel level.

Dynamic or Static Peaks Change the mode of peak indicators. Dynamic Peaks resets the yellow peak level indicators to a new peak level after 1.5 seconds, letting you easily see recent peak amplitude. As the audio gets quieter, the peak indicators recede. Static Peaks retains peak indicators, letting you determine the maximum amplitude of the signal since monitoring, playback, or recording began. However, you can manually reset peak indicators by clicking clip indicators.

***Tip**: To find out how loud audio will get before you record it, choose Static Peaks. Then monitor input levels; the peak indicators show the level of the loudest part.*

Adjust recording levels for standard sound cards

Adjust levels if recordings are too quiet (causing unwanted noise) or too loud (causing distortion). To get the best sounding results, record audio as loud as possible without clipping. When setting recording levels, watch the meters, and try to keep the loudest peaks in the yellow range below -3 dB

Adobe Audition doesn't directly control a sound card's recording levels. For a professional sound card, you adjust these levels with the mixer application provided with the card (see the card's documentation for instructions). For a standard sound card, you use the mixer provided by Windows or Mac OS.

Adjust sound card levels in Windows 7 and Vista

1. Right-click the speaker icon in the taskbar, and choose Recording Devices.
2. Double-click the input source you want to use.
3. Click the Levels tab, and adjust the slider as needed.

Adjust sound card levels in Windows XP

1. Double-click the speaker icon in the taskbar.
2. Choose Options > Properties.
3. Select Recording, and then click OK.
4. Select the input source you want to use, and adjust the Volume slider as needed.

Adjust sound card levels in Mac OS

1. Choose System Preferences from the Apple menu.
2. Click Sound, and then click the Input tab.
3. Select the device you want to use, and adjust the Input Volume slider as needed.

Chapter 5: Applying effects

Enabling CEP extensions

Overview

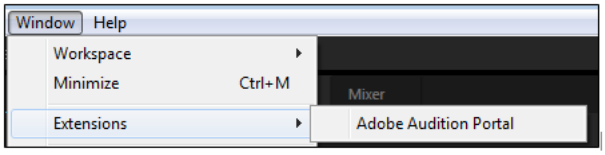
Audition CC supports Common Extensibility Platform (CEP) extensions, which enable developers to build integrated panels in Adobe desktop applications.

Developers design the panels using HTML5 and JavaScript. They use ExtendScript to communicate with the application.

The Adobe-CEP repository at Github contains source code, documentation, and community-built extensions.

Accessing the CEP extensions

Choose **Window** > **Extensions** > **Adobe Audition Portal** to access extensions.



You can install extensions using the Creative Cloud desktop application. Download from the Adobe Add-ons Marketplace and then enable File Sync in the application to automatically install extensions.

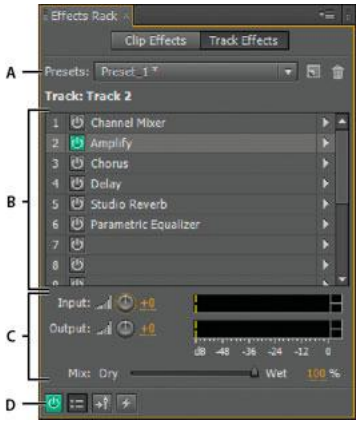
You can use the SDK to create custom CEP extensions. Refer to the documentation and sample code in the SDK.

Download the Audition SDK from the Audition Developer Center.

Effects controls

Effects Rack overview

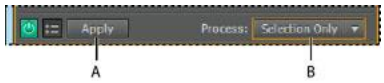
The Effects Rack lets you insert, edit, and reorder up to 16 effects, optimize mix levels, and store favorite presets. Most rack controls appear in both the Waveform and Multitrack editors.



Controls shared by the Waveform and Multitrack editors
***A** Rack Preset controls **B** Effect slots **C** Level controls **D** Main Power button*

Controls unique to the Waveform Editor

In the Waveform Editor, the Effects Rack provides a Process menu. This menu lets you modify a selection or the entire file, and an Apply button that permanently applies effects.



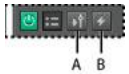
Controls unique to the Waveform Editor
***A** Apply button permanently applies effects **B** Process menu lets you modify selection or entire file*

Controls unique to the Multitrack Editor

The Effects Rack provides Pre-render Track and FX Pre/Post-Fader buttons that you use to optimize and route effects. Each clip and track has its own Effects Rack, which is saved with the session.

Note:

Buses and the Master track lack a Pre-render option because processing effects from all source tracks would reduce performance.



Controls unique to the Multitrack Editor
***A** FX Pre/Post-Fader **B** Pre-render*

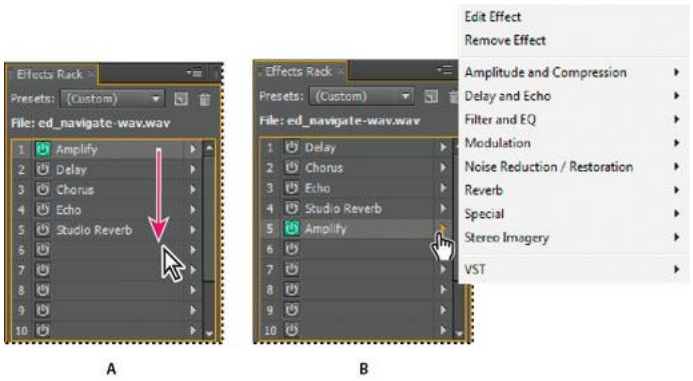
Set input, output, and mix levels in racks

- To optimize volume, adjust Input and Output levels so their meters peak without clipping.
- To change the percentage of processed audio, drag the Mix slider. 100% (Wet) equals fully processed audio; 0% (Dry) equals original, unprocessed audio.

Insert, bypass, reorder, or remove effects in racks

In the Effects Rack, you manage groups of effects by using individual effect slots.

In the Multitrack Editor, the fx section of the Editor panel or Mixer provides quick access to slots in the Effects Rack.



Reordering and inserting effects in racks:
***A** Reorder by dragging **B** Insert with the slot menu*

? To insert an effect, choose it from a slot's pop-up menu. Then adjust effect settings as desired.

To later reaccess effect settings, double-click the effect name in the rack.

? To bypass an effect, click its Power button .

? To bypass all effects, click the main Power button in the lower left corner of a rack. You cab also click the fx power button in the Editor panel or Mixer.

? To bypass a selected group of effects, choose Toggle Power State of Selected Effects from the panel menu .

Bypass effects to quickly compare processed and unprocessed audio.

? To remove a single effect, choose Remove Effect from a slot's pop-up menu. Or select the slot, and press Delete.

? To remove all effects, choose Remove All Effects from the panel menu .

? To reorder effects, drag them to different slots.

Reordering effects produces different sonic results. (For an example, place Reverb before Phaser, or the other way around.)

Use effect presets

Many effects provide presets that let you store and recall favorite settings. In addition to effect-specific presets, the Effects Rack provides rack presets that store groups of effects and settings.

? To apply a preset, choose it from the Presets menu.

1 To save current settings as a preset, click the New Preset button .

? To delete a preset, select it, and click the Delete button .

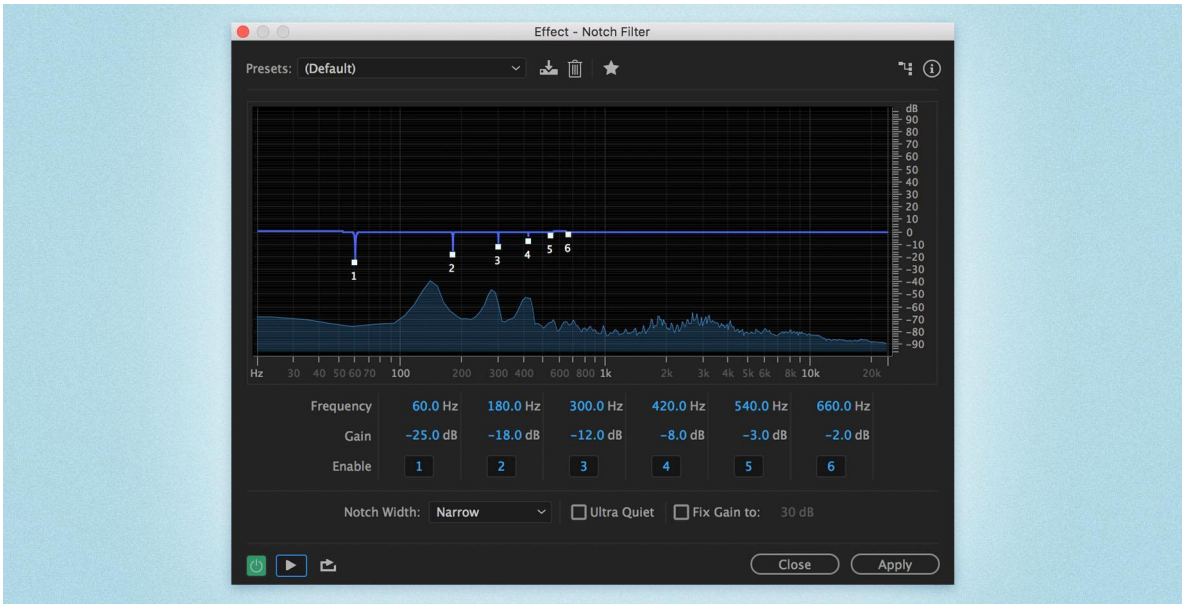
To modify an existing preset, apply it, adjust settings as desired, and then save a new preset with the same name.

Control effect settings with graphs

Many Adobe Audition effects provide graphs where you can adjust parameters. By adding and moving control points on the graph, you can precisely tailor effect settings.

Graph control points function together with related numerical settings. If you change or disable a numerical setting, the related graph control follows suit.

In the effects control graph, you can view the Frequency Spectrum that moves per audio frequency in real time. You can use the frequency spectrum to identify imbalances in audio and use control points to correct them.


Effect - Notch Filter
Presets: (Default)
Frequency 60.0 Hz 180.0 Hz 300.0 Hz 420.0 Hz 540.0 Hz 660.0 Hz
Gain -25.0 dB -18.0 dB -12.0 dB -8.0 dB -3.0 dB -2.0 dB
Enable 1 2 3 4 5 6
Notch Width: Narrow
Ultra Quiet
Fix Gain to: 30 dB
Close
Apply

Frequency Spectrum

? To move a point on a graph, drag it to a new location.

Note:

The following techniques don't apply to the DeHummer, Mastering, Full Reverb, Parametric Equalizer, and Track EQ graphs.

? To add a control point to a graph, click in the grid at the location where you want to place the point.

? To enter numeric values for a control point, right-click it, and choose Edit Point

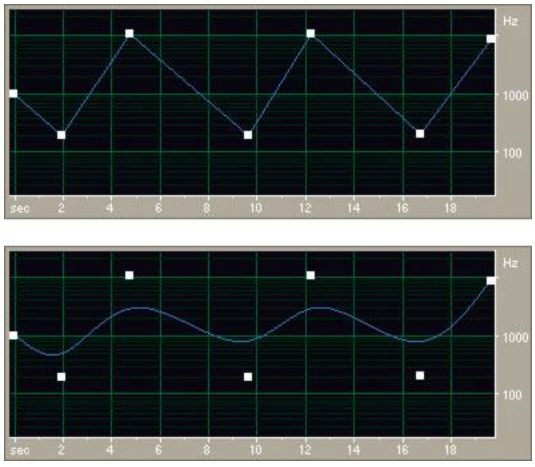
? To remove a point from a graph, drag it off the graph.

? To return a graph to its default state, click the Reset button .

About spline curves for graphs

By default, graphs display straight lines between control points. However, some graphs provide a Spline Curves option that creates a curve between control points for smoother transitions.

When you use spline curves, lines don't travel directly through control points. Instead, the points control the shape of the curve. To move the curve closer to a control point, click near it to create a cluster of control points.



Graph with straight lines compared to graph with spline curves

Applying effects in the Waveform Editor

While previewing effects, you can adjust selections and the current-time indicator in the Editor panel. (The Normalize and Stretch effects are exceptions.)

Apply groups of effects in the Waveform Editor

In the Waveform Editor, the Effects Rack lets you apply groups of effects. (It doesn't provide process effects such as Noise Reduction, which must be applied individually.)

1. Choose Window > Effects Rack.
2. In the numbered list, choose effects for up to 16 slots. (See Insert, bypass, reorder, or remove effects in racks.)
3. Start playback to preview the changes, and then edit, mix, and reorder effects as needed.

   To compare processed audio to original audio, select and deselect the main Power button in the lower left corner of the rack, or the Power buttons for individual effects.
4. To apply the changes to the audio data, click Apply.

To store settings, save a rack preset. (See Use effect presets.)

Apply individual effects in the Waveform Editor

1 From any submenu in the Effects menu, choose an effect.

2 Click the Preview button , and then edit settings as needed.

As you edit settings, watch the Levels panel to optimize amplitude.

3 To compare original audio to processed audio, select and deselect the Power button .

4 To apply the changes to the audio data, click Apply.

About process effects

You can identify process effects by the word *process* in menu commands. These processing-intensive effects are available only offline in the Waveform Editor. Unlike real-time effects, process effects can be applied only individually, so they aren't accessible in the Effects Rack.

Adding third party plugins

Third-party plug-ins let you extend the already powerful effects provided with Adobe Audition. The application supports VST plug-ins on both platforms and Audio Units plug-ins on Mac OS.

Applying plug-in effects is identical to applying built-in effects. For information about plug-in features, consult the documentation provided by the plug-in manufacturer.

Enable VST and Audio Units plug-ins

To access third-party plug-ins in Adobe Audition, you must first enable them. By default, all third-party plug-ins are disabled. To optimize performance, enable only the plug-ins you plan to use in Adobe Audition.

Note:

If effects are being used in a multitrack session, close the session.

1 Choose Effects > Audio Plug-in Manager.

2 In the VST Plug-in Folders section, click Add to specify custom folders you want to scan for plug-ins. Click Default to specify the standard VST folder for your operating system.

3 In the Available Plug-ins section, click Scan For Plug-ins.

If you've recently updated a plug-in, select Rescan Existing Plug-ins.

4 Select the plug-ins you want to access in Adobe Audition, and then click OK.

Note:

If a third-party effect is incompatible, Adobe Audition adds it to an Unsupported submenu in effects menus.

Notch Filter effect

The **Filter and EQ** > **Notch Filter** effect removes up to six user-defined frequency bands. Use this effect to remove very narrow frequency bands, such as a 60 Hz hum, while leaving all surrounding frequencies untouched.

*To remove shrill "ess" sounds, use the **Sibilance Softener** preset. Or use **DTMF** presets to remove standard tones for analog telephone systems.*

Frequency Specifies the center frequency for each notch.

Gain Specifies the amplitude for each notch.

Notch width Determines frequency range for all notches. The three options range from Narrow for a second order filter, which removes some adjacent frequencies, to Super Narrow for a sixth order filter, which is very specific.

Generally, use no more than 30 dB of attenuation for a Narrow setting, 60 dB for Very Narrow, and 90 dB for Super Narrow. Greater attenuation can remove a wide range of neighboring frequencies.

Ultra-Quiet Virtually eliminates noise and artifacts, but requires more processing. This option is audible only on high-end headphones and monitoring systems.

Fix Attenuations To Determines if notches have equal or individual attenuation levels.

Fade and Gain Envelope effects (Waveform Editor only)

The **Amplitude and Compression** > **Fade** and **Gain Envelope** effects function similarly but modify audio differently:

- Choose **Fade Envelope** to reduce amplitude by varying amounts over time.
- Choose **Gain Envelope** to boost or reduce amplitude over time.

In the Editor panel, click the yellow envelope line to add keyframes, and drag them up or down to change amplitude. To quickly select, reposition, or delete multiple keyframes, see Adjust automation with keyframes.

*Select the **Spline Curves** option to create smoother, curved transitions between keyframes, rather than linear transitions. See About spline curves for graphs.*

Manual Pitch Correction effect (Waveform Editor only)

The **Manual Pitch Correction** effect lets you visually adjust pitch with the **Spectral Pitch Display**. The **Spectral Pitch Display** shows the fundamental pitch as a bright blue line, and overtones in yellow to red hues. Corrected pitch appears as a bright green line.

*You can visually monitor pitch at any time, without using the **Manual Pitch Correction** effect. Simply click the **Spectral Pitch Display** icon in the options bar. To customize resolution, decibel range, and gridlines, adjust Pitch Display settings in the Spectral Displays preferences.*

1 Choose **Effects** > **Time and Pitch** > **Manual Pitch Correction**.

2 In the **Manual Pitch Correction** window, set the following options:

Reference Channel Choose a source channel in which pitch changes are most clear. The effect analyzes only the channel you choose, but applies the pitch correction equally to all channels.

Spline Curves Create smoother transitions when using envelope keyframes to apply different pitch correction over time.

Pitch Curve Resolution Sets the Fast Fourier Transform size of each piece of data that the effect processes. In general, use smaller values for correcting higher frequencies. For voices, a setting of 2048 or 4096 sounds most natural, and a setting of 1024 creates robotic effects.

3 In the **Editor** panel, do either of the following:

- To change pitch uniformly, drag the **Adjust Pitch** knob in the heads-up display.
- To change pitch over time, add keyframes to the yellow envelope line in the center of the waveform display.

To zoom in on specific pitch ranges, right-click and drag in the vertical ruler to the right of the Spectral Pitch Display. To reset the zoom level or customize the displayed scale, right-click the ruler and select options from the pop-up menu.

Graphic Phase Shifter effect

The **Stereo Imagery** > **Graphic Phase Shifter** effect lets you adjust the phase of a waveform by adding control points to a graph.

Right-click points to access the Edit Point dialog box for precise, numerical control.

Phase shift graph The horizontal ruler (x-axis) measures frequency, while the vertical ruler (y-axis) displays the degree of phase to shift, where zero is no phase shift. You can create simulated stereo by creating a zigzag pattern that gets more extreme at the high end on one channel.

Frequency Scale Sets the values of the horizontal ruler (x-axis) on a linear or logarithmic scale. Select Logarithmic to work at finer detail in the lower frequencies. (The logarithmic scale better reflects the frequency-emphasis of human hearing.) Select Linear to work at finer detail in the higher frequencies.

Range Sets the values of the vertical ruler (y-axis) on a 360° or 180° scale.

Channel Specifies the channel(s) to apply the phase shift to.

Note:

Process a single channel for the best results. If you apply identical phase shift to two stereo channels, the resulting file sounds exactly the same.

FFT Size Specifies the Fast Fourier Transform size. Higher sizes create more precise results, but they take longer to process.

Doppler Shifter effect (Waveform Editor only)

The **Special** > **Doppler Shifter** effect creates the increase and decrease in pitch we notice when an object approaches and then passes us, such as when a police car passes with its siren on. When the car comes toward you, the sound reaches your ears as a higher frequency because each sound wave is compressed by the car moving forward. The opposite happens as the car passes by; the waves are stretched out, resulting in a lower-pitched sound.

Note:

*Unlike many graphs in Adobe Audition effects, the **Doppler Shifter** graph is noninteractive: You can't directly manipulate the graph. Instead, the graph changes as you adjust the effect's parameters.*

Path Type Defines which path the sound source appears to take. Depending on the path type, a different set of options is available.

Straight Line options:

- **Starting Distance Away** sets the virtual starting point (in meters) of the effect.
- **Velocity** defines the virtual speed (in meters per second) at which the effect moves.
- **Coming From** sets the virtual direction (in degrees) from where the effect appears to come.
- **Passes In Front By** specifies how far (in meters) the effect seems to pass in front of the listener.
- **Passes On Right By** specifies how far (in meters) the effect seems to pass to the right of the listener.

Circular options:

- **Radius** sets the circular dimensions (in meters) of the effect.
- **Velocity** defines the virtual speed (in meters per second) at which the effect moves.
- **Starting Angle** sets the beginning virtual angle (in degrees) of the effect.
- **Center In Front By** specifies how far (in meters) the sound source is from the front of the listener.
- **Center On Right By** specifies how far (in meters) the sound source is from the right of the listener.

Adjust Volume Based on Distance or Direction Automatically adjusts the effect's volume based on the values specified.

Quality Level Provides six different levels of processing quality. Lower quality levels require less processing time, but higher quality levels generally produce better sounding results.

Chapter 6: Effects reference

Apply amplitude and compression effects to audio

Amplify effect

The Amplitude And Compression > Amplify effect boosts or attenuates an audio signal. Because the effect operates in real time, you can combine it with other effects in the Effects Rack.

Gain sliders Boost or attenuate individual audio channels.

Link Sliders Moves the channel sliders together.

Channel Mixer effect

The Amplitude and Compression > Channel Mixer effect alters the balance of stereo or surround channels. You change the apparent position of sounds, correct mismatched levels, or address phasing issues.

Channel tabs Select the output channel.

Input channel sliders To mix into the output channel, determine the percentage of the current channels. For a stereo file, for example, an L value of 50 and an R value of 50 results in an output channel that contains equal audio from the current left and right channels.

Invert Inverts a channel's phase. (To understand this key audio concept, see How sound waves interact.) Inverting all channels causes no perceived difference in sound. Inverting only one channel, however, can greatly change the sound.

DeEsser effect

The Amplitude and Compression > DeEsser effect removes sibilance, "ess" sounds heard in speech and singing that can distort high frequencies.

The graph reveals the processed frequencies. To see how much audio content exists in the processed range, click the Preview button .

Mode Choose Broadband to uniformly compress all frequencies or Multiband to only compress the sibilance range. Multiband is best for most audio content but slightly increases processing time.

Threshold Sets the amplitude above which compression occurs.

Center Frequency Specifies the frequency at which sibilance is most intense. To verify, adjust this setting while playing audio.

Bandwidth Determines the frequency range that triggers the compressor.

Tip To visually adjust Center Frequency and Bandwidth, drag the edges of the selection in the graph.

Output Sibilance Only Lets you hear detected sibilance. Start playback, and fine-tune settings above.

Gain Reduction Shows the compression level of the processed frequencies.

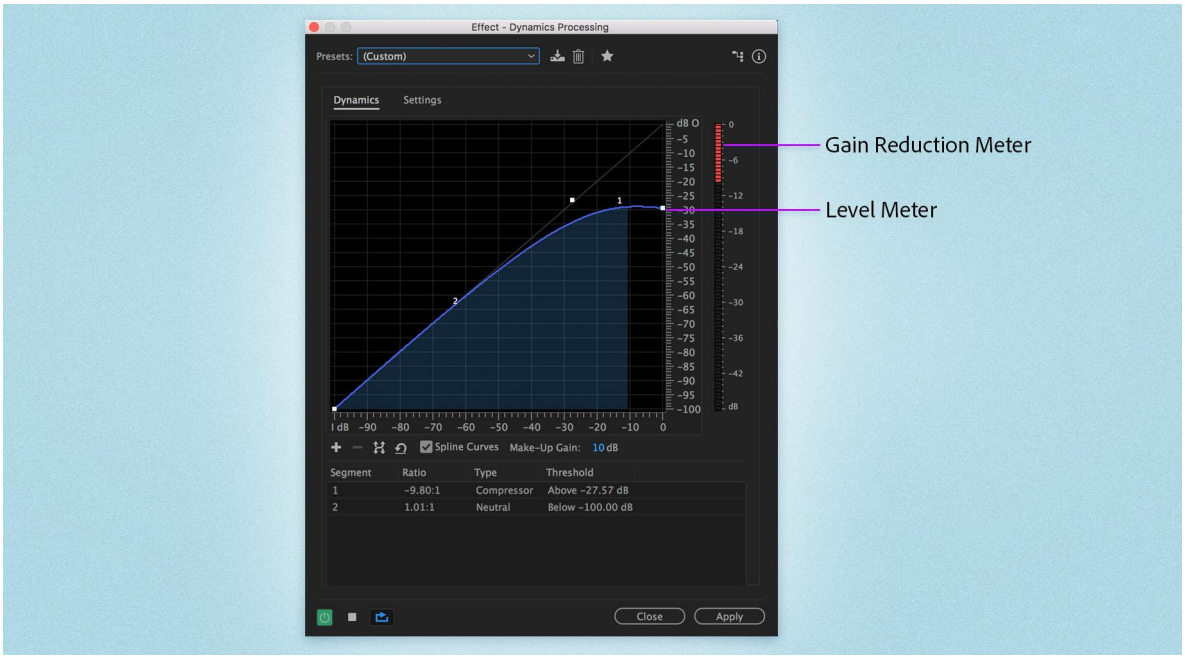
Dynamics Processing effect

The Amplitude And Compression > Dynamics Processing effect can be used as a compressor, limiter, or expander. As a compressor and limiter, this effect reduces dynamic range, producing consistent volume levels. As an expander, it increases dynamic range by reducing the level of low-level signals. (With extreme expander settings, you can create a *noise gate* that totally eliminates noise below a specific amplitude threshold.)

The Dynamics Processing effect can produce subtle changes that you notice only after repeated listening. When applying this effect in the Waveform Editor, use a copy of the original file so you can return to the original audio if necessary.

In the **Dynamic Processing Effect**, you can view the **Level Meter** and the **Gain Reduction Meter.Level Meter** shows the input level of the audio and **Gain Reduction Meter** shows how audio signals are compressed or expanded. These meters are visible on the right side of the graph as shown below.

Use the Broadcast Limiter preset to simulate the processed sound of a contemporary radio station.



Level Meter and Gain Reduction Meter

Dynamics tab

Graph Depicts input level along the horizontal ruler (x-axis) and the new output level along the vertical ruler (y-axis). The default graph, with a straight line from the lower left to the upper right, depicts a signal that has been left untouched; every input level has the same output level. Adjusting the graph changes the relationship between input and output levels, altering dynamic range.

For example, if a desirable sonic element occurs around -20 dB, you can boost the input signal at that level, but leave everything else unchanged. You can also draw an inverse line (from the upper left to the lower right) that boosts quiet sounds and suppress loud ones.

Add point

Adds control point in graph using numerical input and output levels you specify. This method is more precise than clicking the graph to add points.

Tip To numerically adjust an existing control point, right-click it, and choose Edit Point.

Delete point

Removes selected point from the graph.

Invert

Flips the graph, converting compression into expansion, or the other way around.

Note: You can invert a graph only if it has points in the two default corners (-100, -100 and 0, 0) if its output level increases from left to right (that is, each control point must be higher than the one to its left).

Reset

Resets the graph to its default state.

Spline Curves Creates smoother, curved transitions between control points, rather than more abrupt, linear transitions. (See About spline curves for graphs.)

Make-Up Gain Boosts the processed signal.

Settings tab

General Provides overall settings.

Look-Ahead Time Addresses transient spikes that can occur at the onset of loud signals that extend beyond the compressor's Attack Time settings. Extending Look-Ahead Time causes compression to attack before the audio gets loud, ensuring that amplitude never exceeds a certain level. Conversely, reducing Look-Ahead Time is desirable to enhance the impact of percussive music like drum hits.

Noise Gating Completely silences signals that are expanded below a 50-to-1 ratio.

Level Detector Determines the original input amplitude.

Input Gain Applies gain to the signal before it enters the Level Detector.

Attack Time Determines how many milliseconds it takes for the input signal to register a changed amplitude level. For example, if audio suddenly drops 30 dB, the specified attack time passes before the input registers an amplitude change. This avoids erroneous amplitude readings due to temporary changes.

Release Time Determines how many milliseconds the current amplitude level is maintained before another amplitude change can register.

Tip Use fast attack and release settings for audio with fast transients, and slower settings for less percussive audio.

Peak mode Determines levels based on amplitude peaks. This mode is a bit more difficult to use than RMS, because peaks aren't precisely reflected in the Dynamics graph. However, it can be helpful when audio has loud transient peaks you want to subdue.

RMS mode Determines levels based on the root-mean-square formula, an averaging method that more closely matches the way people perceive volume. This mode precisely reflects amplitudes in the Dynamics graph. For example, a limiter (flat horizontal line) at -10 dB reflects an average RMS amplitude of -10 dB.

Gain Processor Amplifies or attenuates the signal depending on the amplitude detected.

Output Gain Applies gain to the output signal after all dynamics processing.

Attack Time Determines how many milliseconds it takes for the output signal to reach the specified level. For example, if audio suddenly drops 30 dB, the specified attack time passes before the output level changes.

Release Time Determines how many milliseconds the current output level is maintained.

Note: If the sum of Attack and Release times is too short (less than about 30 milliseconds), audible artifacts can be heard. To see good attack and release times for different types of audio content, choose various options from the Presets menu.

Link Channels Processes all channels equally, preserving the stereo or surround balance. For example, a compressed drum beat on the left channel reduces the right channel level by an equal amount.

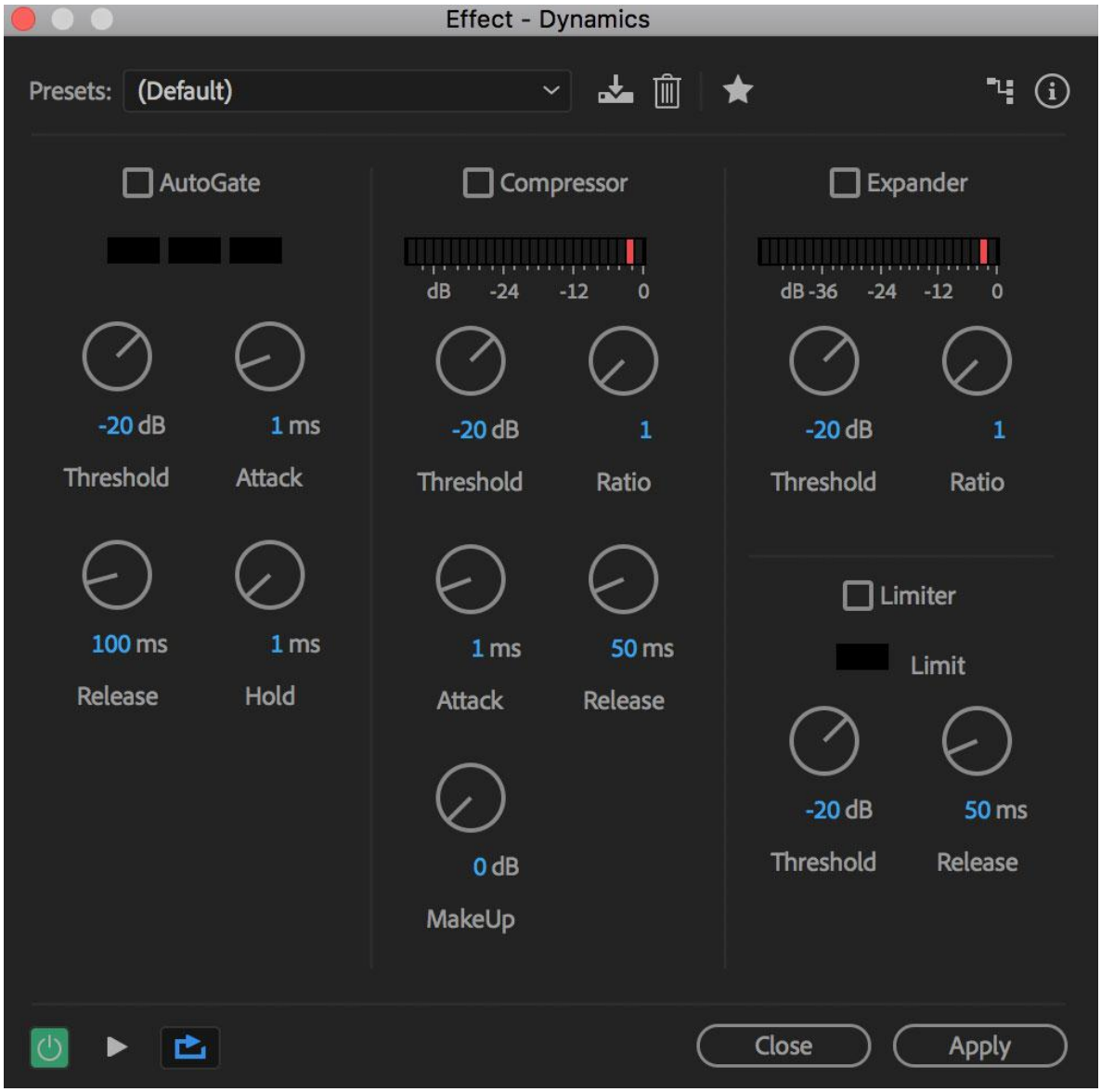
Band Limiting Restricts dynamics processing to a specific frequency range.

Low Cutoff Is the lowest frequency that dynamics processing affects.

High Cutoff Is the highest frequency that dynamics processing affects.

Dynamic Effects

The Amplitude and Compression > Dynamics Effects consist of four sections. They are Auto Gate, Compressor, Expander, and Limiter. You can individually control each one of the section. The LED and gain reduction meters helps you get the overview about how the audio signal is processed.



Adjust and control Dynamic Effects

The different parameters under Dynamic Effects are as follows:

• **Auto Gate**: Removes noise below a certain amplitude threshold. The LED meter is green when audio passes through the gate. The meter turns red when there is no audio passing, and yellow during the attack, release, and hold times.

• **Compressor**: Reduces the dynamic range of the audio signal by attenuating audio that exceeds a specific threshold. The **Ratio** parameter can be used control the change in dynamic range while **Attack** and **Release** parameter changes the temporal behavior. Use the **Gain** parameter to increase the audio level after compressing the signal. The **Gain Reduction** meter shows how much the audio level is reduced.

• **Expander**: Increases the dynamic range of the audio signal by attenuating audio below the specified threshold. The ratio parameter can be used to control the change in dynamic range. The gain reduction meter shows the level of reduction in audio level.

• **Limiter**: Attenuate audio that exceeds a specified threshold. The meter LED turns on when the signal is limited.

Fade Envelope effect

To reduce amplitude by varying amounts over time, choose **Fade Envelope** (**Effects > Amplitude and Compression**).

In the Waveform Editor panel, click the yellow envelope line to add keyframes, and drag them up or down to change amplitude. To quickly select, reposition, or delete multiple keyframes, see Adjust automation with keyframes.

*To create smoother, curved transitions between keyframes, rather than linear transition, select the **Spline Curves** option. See About spline curves for graphs.*

Gain Envelope effect

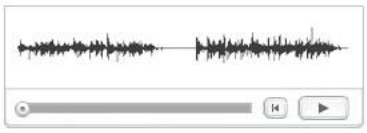
To boost or reduce amplitude over time, choose **Gain Envelope** (**Effects > Amplitude and Compression**).

In the Waveform Editor panel, click the yellow envelope line to add keyframes, and drag them up or down to change amplitude. To quickly select, reposition, or delete multiple keyframes, see Adjust automation with keyframes.

*To create smoother, curved transitions between keyframes, rather than linear transitions, select the **Spline Curves** option t. See About spline curves for graphs.*

Hard Limiter effect

The Amplitude And Compression > Hard Limiter effect greatly attenuates audio that rises above a specified threshold. Typically, limiting is applied with an input boost, a technique that increases overall volume while avoiding distortion.



Maximum Amplitude Sets the maximum sample amplitude allowed.

***Tip**: To avoid clipping when working with 16-bit audio, set this value to no more than -0.3 dB. If you set it even lower, to -3 dB, you'll have a little more clearance for any future edits.*

Input Boost Preamplifies audio before you limit it, making a selection louder without clipping it. As you increase this level, compression increases. Try extreme settings to achieve the loud, high-impact audio heard in contemporary pop music.

Look-Ahead Time Sets the amount of time (in milliseconds) to attenuate the audio before the loudest peak is hit.

Note: Make sure that the value is at least 5 milliseconds. If this value is too small, audible distortion effects occur.

Release Time Sets the time (in milliseconds) for the attenuation to rebound back 12 dB (or roughly the time needed for audio to resume normal volume if a loud peak is encountered). In general, a setting of around 100 (the default) works well and preserves low bass frequencies.

Note: If this value is too large, audio can remain quiet and not resume normal levels for a while.

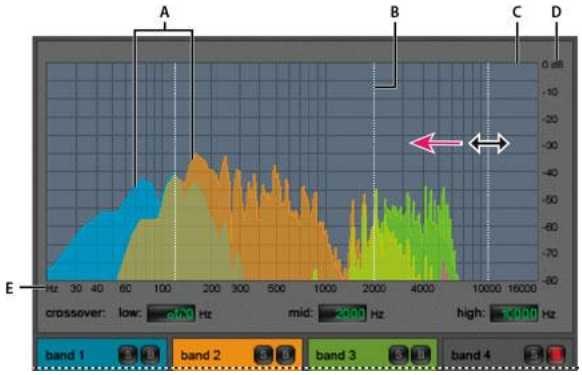
Link Channels Links the loudness of all channels together, preserving the stereo or surround balance.

Multiband Compressor effect

The Amplitude And Compression > Multiband Compressor effect lets you independently compress four different frequency bands. Because each band typically contains unique dynamic content, multiband compression is a powerful tool for audio mastering.

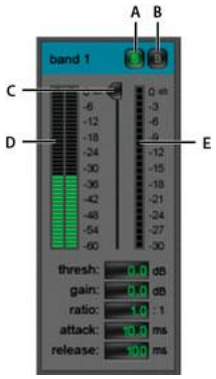
Controls in the Multiband Compressor let you precisely define crossover frequencies and apply band-specific compression settings. To preview bands in isolation, or Bypass buttons to pass bands through without processing, click Solo buttons. After you fine-tune individual bands, select Link Band Controls to adjust them globally, and then optimize overall volume with the Output Gain slider and Limiter settings.

To change compression settings over time, use automation lanes in the Multitrack Editor. (See Automating track settings.)



Adjusting a crossover frequency in the Multiband Compressor
***A** Frequency bands **B** Crossover markers **C** Bypassed band (no processing applied) **D** Amplitude scale **E** Frequency scale*

Crossover Sets the crossover frequencies, which determine the width of each band. Either enter specific Low, Midrange, and High frequencies, or drag the crossover markers above the graph.



Band-specific controls in the Multiband Compressor
***A** Solo **B** Bypass **C** Threshold slider **D** Input Level meters **E** Gain Reduction meters*

Solo buttons Let you hear specific frequency bands. Enable one Solo button at a time to hear bands in isolation, or enable multiple buttons to hear two or more bands together.

Bypass buttons Bypass individual bands so they pass through without processing.

Alt-click (Windows) or Option-click (Mac OS) Solo or Bypass buttons to quickly apply a unique setting to one band.

Threshold sliders Set the input level at which compression begins. Possible values range from -60 to 0 dB. The best setting depends on audio content and musical style. To compress only extreme peaks and retain more dynamic range, try thresholds around 5 dB below the peak input level. To highly compress audio and greatly reduce dynamic range, try settings around 15 dB below the peak input level.

Input Level meters Measure input amplitude. To reset peak and clip indicators, double-click the meters.

Gain Reduction meters Measure amplitude reductions with red meters that extend from top (minimal reduction) to bottom (maximum reduction).

Gain Boosts or cuts amplitude after compression. Possible values range from -18 to +18 dB, where 0 is unity gain.

Ratio Sets a compression ratio between 1-to-1 and 30-to-1. For example, a setting of 3.0 outputs 1 dB for every 3-dB increase above the compression threshold. Typical settings range from 2.0 to 5.0; higher settings produce the compressed sound often heard in pop music.

Attack Determines how quickly compression is applied when audio exceeds the threshold. Possible values range from 0 milliseconds to 500 milliseconds. The default, 10 milliseconds, works well for a wide range of audio. Faster settings work better for audio with fast transients, but such settings sound unnatural for less percussive audio.

Release Determines how quickly compression stops after audio drops below the threshold. Possible values range from 0 milliseconds to 5000 milliseconds. The default, 100 milliseconds, works well for a wide range of audio. Try faster settings for audio with fast transients, and slower settings for less percussive audio.

Output Gain Boosts or cuts overall output level after compression. Possible values range from -18 to +18 dB, where 0 is unity gain. To reset peak and clip indicators, double-click the meters.

Limiter Applies limiting after Output Gain, at the end of the signal path, optimizing overall levels. Specify Threshold, Attack, and Release settings that are less aggressive than similar band-specific settings. Then specify a Margin setting to determine the absolute ceiling relative to 0 dBFS.

Tip: To create compressed audio, enable the Limiter, and then experiment with high Output Gain settings.

Spectrum On Input Displays the frequency spectrum of the input signal, rather than the output signal, in the multiband graph. To quickly see the amount of compression applied to each band, toggle this option on and off.

Brickwall Limiter Applies immediate, hard limiting at the current Margin setting. (Deselect this option to apply slower soft limiting, which sounds less compressed but can exceed the Margin setting.)

Note: The maximum Attack time for brickwall limiting is 5 ms.

Link Band Controls Lets you globally adjust the compression settings for all bands, while retaining relative differences between bands.

Tip To temporarily link band controls, hold down Alt+Shift (Windows) or Option+Shift (Mac OS). To reset a control in all bands, hold down Ctrl+Alt+Shift (Windows) or Command+Option+Shift (Mac OS), and click the control.

Normalize effect (Waveform Editor only)

Note:

This effect requires offline processing. While it is open, you cannot edit the waveform, adjust selections, or move the current-time indicator.

The Amplitude And Compression > Normalize effect lets you set a peak level for a file or selection. When you normalize audio to 100%, you achieve the maximum amplitude that digital audio allows—0 dBFS. If you're sending audio to a mastering engineer, however, normalize audio to between –3 and –6 dBFS, providing a cushion for further processing.

The Normalize effect amplifies the entire file or selection equally. For example, if the original audio reaches a loud peak of 80% and a quiet low of 20%, normalizing to 100% amplifies the loud peak to 100% and the quiet low to 40%.

To apply RMS normalization, choose Effects > Match Volume. If desired, you can apply that command to only one file. (See Match volume across multiple files.)

Normalize To Sets the percentage of the highest peak relative to the maximum amplitude.

Tip: Select dB to enter the Normalize value in decibels instead of a percentage.

Normalize All Channels Equally Uses all channels of a stereo or surround waveform to calculate the amplification amount. If this option is deselected, the amount is calculated separately for each channel, potentially amplifying one considerably more than others.

DC Bias Adjust Lets you adjust the position of the waveform in the wave display. Some recording hardware introduces a DC bias, causing the recorded waveform to appear to be above or below the normal center line in the wave display. To center the waveform, set the percentage to zero. To skew the entire selected waveform above or below the center line, specify a positive or negative percentage.

Single-band Compressor effect

The Amplitude And Compression > Single-band Compressor effect reduces dynamic range, producing consistent volume levels and increasing perceived loudness. Single-band compression is effective for voiceovers, because it helps the speaker stand out over musical soundtracks and background audio.

For examples of highly compressed audio, listen to recordings of modern pop music. By contrast, most jazz recordings are lightly compressed, while typical classical recordings feature no compression at all.

Threshold Sets the input level at which compression begins. The best setting depends on audio content and style. To compress only extreme peaks and retain more dynamic range, try thresholds around 5 dB below the peak input level. To highly compress audio and greatly reduce dynamic range, try settings around 15 dB below the peak input level.

Ratio Sets a compression ratio between 1-to-1 and 30-to-1. For example, a setting of three outputs 1 dB for every 3-dB increase above the threshold. Typical settings range from 2 to 5; higher settings produce the compressed sound often heard in pop music.

Attack Determines how quickly compression starts after audio exceeds the Threshold setting. The default, 10 milliseconds, works well for a wide range of source material. Use faster settings only for audio with quick transients, such as percussion recordings.

Release Determines how quickly compression stops when audio drops below the Threshold setting. The default, 100 milliseconds, works well for a wide range of audio. Try faster settings for audio with fast transients, and slower settings for less percussive audio.

Output Gain Boosts or cuts amplitude after compression. Possible values range from -30 dB to +30 dB, where 0 is unity gain.

Speech Volume Leveler effect

The Amplitude and Compression > Speech Volume Leveler is a compression effect that optimizes dialogue, evening out levels and removing background noise.

For the best results, do the following:

1. Select audio with the lowest level. Set Target Volume Level and Leveling Amount to the left. Start playback, and gradually increase the Leveling Amount until speech becomes nicely audible without increasing background noise.
2. Select audio with the highest level, and start playback. Adjust the Target Volume Level until the volume matches the loudness of the quiet passage you adjusted previously.
3. If necessary, readjust the Leveling Amount to avoid an over-compressed sound.

Here are more details about each option:

Target Volume Level Sets the desired output level relative to zero dBFS. (See Measuring amplitude in dBFS.)

Leveling Amount At low settings, amplifies speech slightly without boosting the noise floor. At high settings, amplifies the entire signal more as the signal drops closer to the noise floor.

Boost Low Signals Interprets shorter, low-volume passages as speech that you want to amplify. For most audio content, deselect this option to produce smoother sound.

Target Dynamic Range Minimizes background noises while amplifying and leveling only speech.

For best results with most content, specify a Target Dynamic Range value between –45dB and –60dB.

Note:

Specifying a higher parameter value (up to –80dB) can significantly reduce background noise, but applies less leveling and amplification of speech content.

Specifying a lower parameter value (down to –30dB) provides better speech leveling, but can amplify the background noise more than desired.

Advanced settings To access the following options, click the triangle:

Compressor Maintains a strong level if the processed signal falls below a threshold relative to zero dBFS.

Noise Gate Eliminates background noise by dramatically reducing output level when the signal drops by an offset you specify.

Tube-modeled Compressor effect

The Amplitude And Compression > Tube-modeled Compressor effect simulates the warmth of vintage hardware compressors. Use this effect to add subtle distortion that pleasantly colors audio.

Threshold slider Sets the input level at which compression begins. Possible values range from -60 to 0 dB. The best setting depends on audio content and musical style. To compress only extreme peaks and retain more dynamic range, try thresholds around 5 dB below the peak input level; to highly compress audio and greatly reduce dynamic range, try settings around 15 dB below the peak input level.

Input Level meters To the left of the slider, these meters measure input amplitude. To reset peak and clip indicators, double-click the meters.

Gain Reduction meters To the right of the slider, these meters measure amplitude reduction with red bars that extend from top (minimal reduction) to bottom (maximum reduction).

Gain Boosts or cuts amplitude after compression. Possible values range from -18 to +18 dB, where 0 is unity gain.

Ratio Sets a compression ratio between 1-to-1 and 30-to-1. For example, a setting of 3.0 outputs 1 dB for every 3 dB increase above the compression threshold. Typical settings range from 2.0 to 5.0; higher settings produce the compressed sound often heard in pop music.

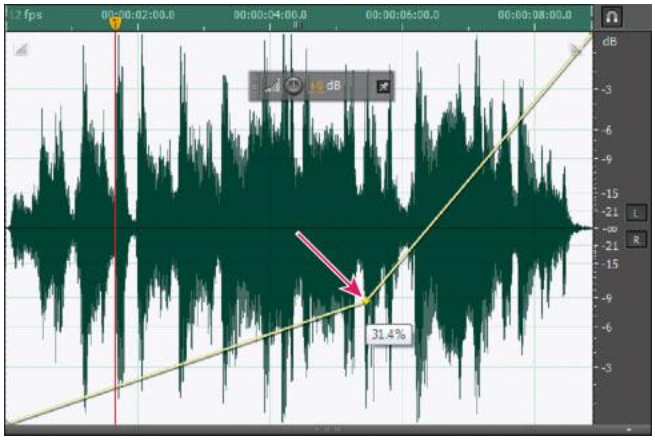
Attack Determines how quickly compression is applied when audio exceeds the threshold. Possible values range from 0 to 500 milliseconds. The default, 10 milliseconds, works well for a wide range of audio. Faster settings work better for audio with fast transients, but such settings sound unnatural for less percussive audio.

Release Determines how quickly compression stops after audio drops below the threshold. Possible values range from 0 to 5000 milliseconds. The default, 100 milliseconds, works well for a wide range of audio. Try faster settings for audio with fast transients, and slower settings for less percussive audio.

Volume Envelope effect

The **Amplitude And Compression > Volume Envelope** effect lets you change volume over time with boosts and fades. In the Waveform Editor panel, simply drag the yellow line. The top of the panel represents 100% (normal) amplification; the bottom represents 100% attenuation (silence).

Though the Volume Envelope effect isn't available in the Multitrack Editor, you can use automation lanes to accomplish the same task. (See Automating track settings.)



Dragging an anchor point in the Editor panel

Yellow envelope line in Editor panel Drag to adjust amplitude percentage, and click to add keyframes for more boosts and fades. To quickly select, reposition, or delete multiple keyframes, see Adjust automation with keyframes.

Spline curves Applies smoother, curved transitions between keyframes, rather than linear transitions. See About spline curves for graphs.

Delay and echo effects

Delays are separate copies of an original signal that reoccur within milliseconds of each other. *Echoes* are sounds that are delayed far enough in time so that you hear each as a distinct copy of the original sound. When reverb or chorus might muddy the mix, both delays and echoes are a great way to add ambience to a track.

To access familiar options from hardware delays, use the Echo effect in Adobe Audition.

Analog Delay effect

The Delay And Echo > Analog Delay effect simulates the sonic warmth of vintage hardware delay units. Unique options apply characteristic distortion and adjust the stereo spread. To create discrete echoes, specify delay times of 35 milliseconds or more; to create more subtle effects, specify shorter times.

Mode Specifies the type of hardware emulation, determining equalization and distortion characteristics. Tape and Tube reflect the sonic character of vintage delay units, while Analog reflects later electronic delay lines.

Dry Out Determines the level of original, unprocessed audio.

Wet Out Determines the level of delayed, processed audio.

Delay Specifies the delay length in milliseconds.

Feedback Creates repeating echoes by resending delayed audio through the delay line. For example, a setting of 20% sends delayed audio at one-fifth of its original volume, creating echoes that gently fade away. A setting of 200% sends delayed audio at double its original volume, creating echoes that quickly grow in intensity.

***note**: When experimenting with extremely high Feedback settings, turn down your system volume.*

Trash Increases distortion and boosts low frequencies, adding warmth.

Spread Determines the stereo width of the delayed signal.

Delay effect

The Delay And Echo > Delay effect can be used to create single echoes, as well as a number of other effects. Delays of 35 milliseconds or more create discrete echoes, while those between 15-34 milliseconds can create a simple chorus or flanging effect. (These results won't be as effective as the Chorus or Flanger effects in Adobe Audition, because the delay settings don't change over time.)

By further reducing a delay to between 1 and 14 milliseconds, you can spatially locate a mono sound so that the sound seems to be coming from the left or the right side, even though the actual volume levels for left and right are identical.

Delay Time Adjusts the delay for both the left and right channels from -500 milliseconds to +500 milliseconds. Entering a negative number means that you can move a channel ahead in time instead of delaying it. For instance, if you enter 200 milliseconds for the left channel, the delayed portion of the affected waveform is heard before the original part.

Mix Sets the ratio of processed, Wet signal to original, Dry signal to be mixed into the final output. A value of 50 mixes the two evenly.

Invert Inverts the phase of the delayed signal, creating phase-cancellation effects similar to comb filters. (To understand phase cancellation, see How sound waves interact.)

Echo effect

The Delay And Echo > Echo effect adds a series of repeated, decaying echoes to a sound. (For a single echo, use the Delay effect instead.) You can create effects ranging from a Grand Canyon-type "Hello-ello-llo-lo-o" to metallic, clanging drainpipe sounds by varying the delay amount. By equalizing the delays, you can change a room's characteristic sound from one with reflective surfaces (creating echoes that sound brighter) to one that is almost totally absorptive (creating echoes that sound darker).

Note:

Make sure the audio file is long enough for the echo to end. If the echo is cut off abruptly before it fully decays, undo the Echo effect, add several seconds of silence by choosing Generate > Silence, and then reapply the effect.

Delay Time Specifies the number of milliseconds, beats, or samples between each echo. For example, a setting of 100 milliseconds results in a 1/10th-second delay between successive echoes.

Feedback Determines the falloff ratio of an echo. Each successive echo tails off at a certain percentage less than the previous one. A decay setting of 0% results in no echo at all, while a decay of 100% produces an echo that never gets quieter.

Echo Level Sets the percentage of echoed (wet) signal to mix with the original (dry) signal in the final output.

***Tip**: You can create striking stereo echo effects by setting different left and right values for the Delay Time, Feedback, and Echo Level controls.*

Lock Left & Right Links the sliders for Decay, Delay, and Initial Echo Volume, maintaining the same settings for each channel.

Echo Bounce Makes the echoes bounce back and forth between the left and right channels. If you want to create one echo that bounces back and forth, select an initial echo volume of 100% for one channel and 0% for the other. Otherwise, the settings for each channel will bounce to the other, creating two sets of echoes on each channel.

Successive Echo Equalization Passes each successive echo through an eight-band equalizer, letting you simulate the natural sound absorption of a room. A setting of 0 leaves the frequency band unchanged, while a maximum setting of -15 decreases that frequency by 15 dB. And, because -15 dB is the difference of each successive echo, some frequencies will die out much faster than others.

Delay Time Units Specifies milliseconds, beats, or samples for the Delay Time setting.

Diagnostics effects (Waveform Editor only) for Audition

Diagnostics are available either via the Effects menu or directly from the Diagnostics panel (Window > Diagnostics). These tools let you quickly remove clicks, distortion, or silence from audio, as well as add markers where silence occurs.

For maximum audio restoration control, use diagnostics together with Spectral Display tools and Noise Reduction effects. See Techniques for restoring audio.

Diagnose and repair, delete, or mark audio

Unlike conventional noise reduction effects, which process all selected audio, diagnostics scan for problematic or silent areas, and then let you choose which to address.

1. In the Diagnostics panel, choose an option from the Effect menu.
2. Click Scan.
3. At the bottom of the panel, do any of the following:
   - Select one or more detected items in the list, and click Repair, Delete, or Mark. (The available options depend upon the chosen diagnostic effect.)

     Note: To mark detected clicks or clipping, right-click selected items in the list, and choose Create Markers from the pop-up menu. (See Working with markers.)
   - Click Repair All, Delete All, or Mark All to address all detected items.
   - Click the magnifying glass to zoom in on a selected problem in the Editor panel. Click the icon again to zoom out.
   - Click Clear Repaired, Deleted, or Marked to remove previously addressed items from the list.

DeClicker options

The Diagnostics > DeClicker effect detects and removes clicks and pops from wireless microphones, vinyl records, and other sources.

DeClicker options match those for the Automatic Click Remover, which you can combine with other effects in the Effects Rack and apply in the Multitrack Editor. (See Automatic Click Remover effect.) That effect also applies multiple scan and repair passes automatically; to achieve the same level of click reduction with the DeClicker, you must manually apply it multiple times. However, the DeClicker lets you evaluate detected clicks and choose which to address.

In the Diagnostics panel, click Settings to access these options:

Threshold Determines sensitivity to noise. Lower settings detect more clicks and pops but may include audio you wish to retain. Settings range from 1 to 100; the default is 30.

Complexity Indicates the complexity of noise. Higher settings apply more processing but can degrade audio quality. Settings range from 1 to 100; the default is 16.

To visually identify clicks, zoom in and use Spectral Frequency Display with a resolution of 256 bands. (You can access this setting in the Spectral Display tab of the Preferences dialog box.) Most clicks appear as bright vertical bars that extend from the top to the bottom of the display.

DeClipper options

The Diagnostics > DeClipper effect repairs clipped waveforms by filling in clipped sections with new audio data. Clipping occurs when audio amplitude exceeds the maximum level for the current bit depth. Commonly, clipping results from recording levels that are too high. You can monitor clipping during recording or playback by watching the Level Meters; when clipping occurs, the boxes on the far right of the meters turn red.

Visually, clipped audio appears as broad flat areas at the top of a waveform. Sonically, clipped audio is a static-like distortion.

Note:

If you need to adjust the DC offset of clipped audio, first use the DeClipper effect. If you instead adjust DC offset first, the DeClipper won't identify clipped areas that fall below 0 dBFS.

In the Diagnostics panel, click Settings to access these options:

Gain Specifies the amount of attenuation that occurs before processing. Click Auto to base the gain setting on average input amplitude.

Tolerance Specifies the amplitude variation in clipped regions. A value of 0% detects clipping only in perfectly horizontal lines at maximum amplitude; 1% detects clipping beginning at 1% below maximum amplitude, and so on. (A value of 1% detects most clipping.)

Min. Clip Size Specifies the length of the shortest run of clipped samples to repair. Lower values repair a higher percentage of clipped samples; higher values repair clipped samples only if they're preceded or followed other clipped samples.

Interpolation The Cubic option uses spline curves to re-create the frequency content of clipped audio. This approach is faster for most situations but can introduce spurious new frequencies. The FFT option uses Fast Fourier transforms to re-create clipped audio. This approach is typically slower but best for severe clipping. From the FFT Size menu, choose the number of frequency bands to evaluate and replace. (More bands result in greater accuracy but longer processing.)

To retain amplitude when restoring clipped audio, apply the DeClipper effect with a Gain setting of zero, followed by the Hard Limiting effect with a Boost value of zero and a Limit value of -0.2 dB.

Delete Silence and Mark Audio options

The Diagnostics > Delete Silence and Mark Audio effects identify silent passages of audio and either remove or mark them. (See Working with markers.) Automatically deleting silence helps you tighten up tracks without affecting foreground audio. Automatically marking silence helps you quickly navigate to audio cues for editing.

In the Diagnostics panel, click Settings to access these options:

Define Silence As Specifies the amplitude and duration identified as silence.

Define Audio As Specifies the amplitude and duration identified as audio content.

Find Levels Automatically calculates the signal levels of silence and audio based on content in the file.

Fix By (Delete Silence only) Choose Shortening Silence to reduce silent passages to the specified number of milliseconds. Choose Deleting Silence to mute silent passages but retain file length. (Deleting silence helps maintain video synchronization with audio clips in video editing applications.)

To divide sections of sound or speech separated by silence into different files, apply the Mark Audio effect, and click Mark All. Then see Save audio between markers to new files.

Filter and equalizer effects

FFT Filter effect

The graphic nature of the **Filter And EQ** > **FFT Filter** effect makes it easy to draw curves or notches that reject or boost specific frequencies. **FFT** stands for **Fast Fourier Transform**, an algorithm that quickly analyzes frequency and amplitude.

This effect can produce broad high- or low-pass filters (to maintain high or low frequencies), narrow band-pass filters (to simulate the sound of a telephone call), or notch filters (to eliminate small, precise frequency bands).

Scale Determines how frequencies are arranged along the horizontal x-axis:

- For finer control over low frequencies, select **Logarithmic**. A logarithmic scale more closely resembles how people hear sound.
- For detailed, high-frequency work with evenly spaced intervals in frequency, select **Linear**.

Spline Curves Creates smoother, curved transitions between control points, rather than more abrupt, linear transitions. (See About spline curves for graphs.)

Reset

Reverts the graph to the default state, removing filtering.

Advanced options Click the triangle to access these settings:

FFT Size Specifies the **Fast Fourier Transform** size, determining the tradeoff between frequency and time accuracy. For steep, precise frequency filters, choose higher values. For reduced transient artifacts in percussive audio, choose lower values. Values between 1024 and 8192 work well for most material.

Window Determines the **Fast Fourier Transform** shape, with each option resulting in a different frequency response curve.

These functions are listed in order from narrowest to widest. Narrower functions include fewer surrounding, or sidelobe, frequencies but less precisely reflect center frequencies. Wider functions include more surrounding frequencies but more precisely reflect center frequencies. The **Hamming** and **Blackman** options provide excellent overall results.

Graphic Equalizer effect

The **Filter And EQ** > **Graphic Equalizer** effect boosts or cuts specific frequency bands and provides a visual representation of the resulting EQ curve. Unlike the **Parametric Equalizer**, the **Graphic Equalizer** uses preset frequency bands for quick and easy equalization.

You can space frequency bands at the following intervals:

- One octave (10 bands)
- One-half octave (20 bands)
- One-third octave (30 bands)

Graphic equalizers with fewer bands provide quicker adjustment; more bands provide greater precision.

Gain sliders Sets the exact boost or attenuation (measured in decibels) for the chosen band.

Range Defines the range of the slider controls. Enter any value between 1.5 and 120 dB. (By comparison, standard hardware equalizers have a range of about 12 to 30 dB.)

Accuracy Sets the accuracy level for equalization. Higher accuracy levels give better frequency response in the lower ranges, but they require more processing time. If you equalize only higher frequencies, you can use lower accuracy levels.

*If you equalize extremely low frequencies, set **Accuracy** to between 500 and 5000 points.*

Master Gain Compensates for an overall volume level that is too soft or too loud after the EQ settings are adjusted. The default value of 0 dB represents no master gain adjustment.

Note:

*The **Graphic Equalizer** is an FIR (Finite Impulse Response) filter. FIR filters better maintain phase accuracy but have slightly less frequency accuracy than IIR (Infinite Impulse Response) filters like the **Parametric Equalizer**.*

Notch Filter effect

The **Filter and EQ** > **Notch Filter** effect removes up to six user-defined frequency bands. Use this effect to remove very narrow frequency bands, such as a 60 Hz hum, while leaving all surrounding frequencies untouched.

*To remove shrill "ess" sounds, use the **Sibilance Softener** preset. Or use **DTMF** presets to remove standard tones for analog telephone systems.*

Frequency Specifies the center frequency for each notch.

Gain Specifies the amplitude for each notch.

Notch width Determines frequency range for all notches. The three options range from **Narrow** for a second order filter, which removes some adjacent frequencies, to **Super Narrow** for a sixth order filter, which is very specific.

*Generally, use no more than 30 dB of attenuation for a **Narrow** setting, 60 dB for **Very Narrow**, and 90 dB for **Super Narrow**. Greater attenuation can remove a wide range of neighboring frequencies.*

Ultra-Quiet Virtually eliminates noise and artifacts, but requires more processing. This option is audible only on high-end headphones and monitoring systems.

Fix Attenuations To Determines if notches have equal or individual attenuation levels.

Parametric Equalizer effect

The **Filter And EQ** > **Parametric Equalizer** effect provides maximum control over tonal equalization. Unlike the **Graphic Equalizer**, which provides a fixed number of frequencies and Q bandwidths, the **Parametric Equalizer** gives you total control over frequency, Q, and gain settings. For example, you can simultaneously reduce a small range of frequencies centered around 1000 Hz, boost a broad low-frequency shelf centered around 80 Hz, and insert a 60 Hz notch filter.

The **Parametric Equalizer** uses second-order IIR (Infinite Impulse Response) filters, which are very fast and provide very accurate frequency resolution. For example, you can precisely boost a range of 40 to 45 Hz. FIR (Finite Impulse Response) filters like the **Graphic Equalizer** provide slightly improved phase accuracy, however.

Master Gain Compensates for an overall volume level that's too loud or too soft after you adjust the EQ settings.

Graph Shows frequency along the horizontal ruler (*x*-axis) and amplitude along the vertical ruler (*y*-axis). Frequencies in the graph range from lowest to highest in a logarithmic fashion (evenly spaced by octaves).

Identifying band-pass and shelving filters in the Parametric Equalizer:
***A** High- and low-pass filters **B** High and low shelving filters*

Frequency Sets the center frequency for bands 1-5, and the corner frequencies for the band-pass and shelving filters.

Use the low shelving filter to reduce low-end rumble, hum, or other unwanted low-frequency sounds. Use the high shelving filter to reduce hiss, amplifier noise, and the like.

Gain Sets the boost or attenuation for frequency bands, and the per-octave slope of the band-pass filters.

Q / Width Controls the width of the affected frequency band. Low Q values affect a larger range of frequencies. Very high Q values (close to 100) affect a very narrow band and are ideal for notch filters removing particular frequencies, like 60 Hz hum.

When a very narrow band is boosted, audio tends to ring or resonate at that frequency. Q values of 1-10 are best for general equalization.

Band Enables up to five intermediate bands, as well as high-pass, low-pass, and shelving filters, giving you very fine control over the equalization curve. Click the band button to activate the corresponding settings above.

The low and high shelving filters provide slope buttons ( ) that adjust the low and high shelves by 12 dB per octave, rather than the default 6 dB per octave.

To visually adjust enabled bands in the graph, drag the related control points.

Constant Q, Constant Width Describes a frequency band's width as either a Q value (which is a ratio of width to center frequency) or an absolute width value in Hz. Constant Q is the most common setting.

Ultra-Quiet Virtually eliminates noise and artifacts, but requires more processing. This option is audible only on high-end headphones and monitoring systems.

Range Sets the graph to a 30 dB range for more precise adjustments, or a 96 dB range for more extreme adjustments.

Scientific Filter effect

Use the **Scientific Filter** effect (**Effects > Filter and EQ**) for advanced manipulation of audio. You can also access the effect from the **Effects Rack** for single assets in the waveform editor, or for tracks and clips in the **Multitrack editor**.

Types Specifies the type of scientific filter. The available options are as follows.

Bessel Provides accurate phase response with no ringing or overshoot. However, the pass band slopes at its edges, where rejection of the stop band is the poorest of all filter types. These qualities make **Bessel** a good choice for percussive, pulse-like signals. For other filtering tasks, use **Butterworth**.

Butterworth Provides a flat pass band with minimal phase shift, ringing, and overshoot. This filter type also rejects the stop band much better than **Bessel** and only slightly worse than **Chebychev** 1 or 2. These overall qualities make **Butterworth** the best choice for most filtering tasks.

Chebychev Provides the best stop band rejection but the worst phase response, ringing, and overshoot in the pass band. Use this filter type only if rejecting the stop band is more important than maintaining an accurate pass band.

Elliptical Provides a sharp cut-off and narrow transition width. It can also notch out frequencies, unlike the Butterworth and Chebychev filters. It can, however,introduce ripples in both the stop band and the pass band

Modes Specify a mode for the filter. The available options are as follows.

Low Pass Passes the low frequencies and removes high frequencies. You must specify the cutoff point at which the frequencies are removed.

High Pass Passes high frequencies and removes low frequencies. You must specify the cutoff point at which the frequencies are removed.

Band Pass Preserves a band, a range of frequencies, while removing all other frequencies. You must specify two cutoff points to define the edges of the band.

Band Stop Rejects any frequencies within the specified range. Also known as a notch filter, **Band Stop** is the opposite of **Band Pass**. You must specify two cutoff points to define the edges of the band.

Master Gain Compensates for an overall volume level that might be too loud or too soft after you adjust the filter settings.

Cutoff Defines the frequency that serves as a border between passed and removed frequencies. At this point the filter switches from passing to attenuating, or vice versa. In filters requiring a range (**Band Pass** and **Band Stop**), Cutoff defines the low frequency border, while **High Cutoff** defines the high frequency border.

High Cutoff Defines the high frequency border in filters that require a range (**Band Pass** and **Band Stop**).

Order Determines the filter's precision. The higher the order, the more precise the filter (with steeper slopes at the cutoff points, and so on). However, very high orders can also have high levels of phase distortion.

Transition Bandwidth (**Butterworth** and **Chebychev** only) Sets the width of the transition band. (Lower values have steeper slopes.) If you specify a transition bandwidth, the **Order** setting is filled in automatically, and vice versa. In filters that require a range (**Band Pass** and **Band Stop**), this serves as the lower frequency transition, while **High Width** defines the higher frequency transition.

High Width (**Butterworth** and **Chebychev** only) In filters that require a range (**Band Pass** and **Band Stop**), this option serves as the higher frequency transition, while **Transition Bandwidth** defines the lower frequency transition.

Stop Attn (**Butterworth** and **Chebychev** only) Determines how much gain reduction to use when frequencies are removed.

Pass Ripple/ Actual Ripple (**Chebychev** only) Determines the maximum allowable amount of ripple. Ripple is the effect of unwanted boosting and cutting of frequencies near the cutoff point.

Modulation effects

Chorus effect

The Modulation > Chorus effect simulates several voices or instruments played at once by adding multiple short delays with a small amount of feedback. The result is lush, rich sound. You can use Chorus to enhance a vocal track or add stereo spaciousness to mono audio.

Adobe Audition uses a direct-simulation method to achieve a chorus effect, making each voice sound distinct from the original by slightly varying timing, intonation, and vibrato. The Feedback setting lets you add extra detail to the result.

To achieve the best results with mono files, convert them to stereo before applying the Chorus effect.

Characteristics Represent the characteristics of each voice in the chorus.

Voices Determines the number of simulated voices.

***Note**: As you add more voices, the sound becomes richer and richer —but processing time also increases.*

Delay Time Specifies the maximum amount of delay allowed. Chorusing introduces short delays (often in the 15-35 millisecond range) that vary in duration over time. If the setting is very small, all the voices start merging into the original, and an unnatural flanging effect might occur. If the setting is too high, a warbled effect might occur, like a tape being eaten by a cassette deck.

Delay Rate Determines how quickly the delay cycles from zero to the maximum delay setting. Because the delay varies over time, the pitch of the sample increases or decreases over time, giving the effect of separate, slightly out of tune voices. For example, a rate of 2 Hz would vary the delay from zero to the maximum and back twice per second (simulating a pitch vibrato at twice per second). If this setting is too low, the individual voices don't vary much in pitch. If it is set too high, the voices may vary so quickly that a warbled effect might occur.

Feedback Adds a percentage of processed voices back into the effect input. Feedback can give a waveform an extra echo or reverb effect. A little feedback (less than 10%) can provide extra richness, depending on the delay and vibrato settings. Higher settings produce more traditional feedback, a loud ringing which can get loud enough to clip the signal.

Spread Gives an added delay to each voice, separating them in time by as much as 200 milliseconds (1/5th of a second). High values cause the separate voices to start at different times—the higher the value, the farther apart the onset of each voice may be. In contrast, low values cause all voices to be in unison. Depending on other settings, low values can also produce flanging effects, which may be undesirable if your goal is a realistic chorus effect.

Modulation Depth Determines the maximum variation in amplitude that occurs. For example, you can alter the amplitude of a chorused voice so that it is 5 dB louder or quieter than the original. At extremely high settings, the sound may cut in and out, creating an objectionable warble. At extremely low settings (less than 1 dB), the depth may be unnoticeable unless the Modulation Rate is set extremely high. Natural vibratos occur around 2 dB to 5 dB.

Note that this setting is a maximum only; the vibrato volume might not always go as low as the setting indicates. This limitation is intentional, as it creates a more natural sound.

Modulation Rate Determines the maximum rate at which amplitude changes occur. With very low values, the resulting voice slowly gets louder and quieter, like a singer that cannot keep his or her breath steady. With very high settings, the result can be jittery and unnatural.

Highest Quality Ensures the best quality results. Increasing the quality, however, increases the processing time for previewing and applying the effect.

Stereo Width Determines where the individual voices are placed in the stereo field and how the original stereo signal is interpreted. These options are active only when you work with stereo files:

Average Left & Right Channel Input Combines the original left and right channels. If deselected, the channels are kept separate to preserve the stereo image. Leave this option deselected if the stereo source audio was originally monophonic—it won't have any effect other than increasing processing time.

Add Binaural Cues Adds separate delays to the left and right outputs of each voice. This delay can make each voice seem to come from a different direction when you listen through headphones. For greater stereo separation, deselect this option for audio that will be played through standard speakers .

Stereo Field Specifies where chorused voices are placed across the left and right stereo image. At lower settings, voices are closer to the center of the stereo image. At a setting of 50%, voices are spaced evenly from left to right. At higher settings, voices move to the outer edges. If you use an odd number of voices, one is always directly in the center.

Output Level Sets the ratio of original (Dry) signal to chorused (Wet) signal. Extremely high settings may cause clipping.

In the Multitrack Editor, you can vary the Wet level over time with automation lanes. (See Automating track settings.)This technique is handy for emphasizing vocal or instrumental solos.

Chorus/Flanger effect

The Modulation > Chorus/Flanger effect combines two popular delay-based effects. The Chorus option simulates several voices or instruments played at once by adding multiple short delays with a small amount of feedback. The result is lush, rich sound. Use this effect to enhance vocal tracks or add stereo spaciousness to mono audio.

The Flanger option creates a psychedelic, phase-shifted sound by mixing a varying, short delay with the original signal. This effect was originally created by sending an identical audio signal to two reel-to-reel tape recorders, and periodically pressing the flange of one reel to slow it down.

Chorus Simulates several voices or instruments playing at once.

Flanger Simulates the delayed, phase-shifted sound originally heard in psychedelic music.

Speed Controls the rate at which the delay time cycles from zero to the maximum setting.

Width Specifies the maximum amount of delay.

Intensity Controls the ratio of original to processed audio.

Transience Emphasizes transients, giving them a sharper, more distinct sound.

Flanger effect

Flanging is an audio effect caused by mixing a varying, short delay in roughly equal proportion to the original signal. It was originally achieved by sending an identical audio signal to two reel-to-reel tape recorders, and then pressing the flange of one reel to slow it down. Combining the two resulting recordings produced a phase-shifted, time-delay effect, characteristic of psychedelic music of the 1960s and 1970s. The Modulation > Flanger effect lets you create a similar result by slightly delaying and phasing a signal at specific or random intervals.

Initial Delay Time Sets the point in milliseconds at which flanging starts behind the original signal. The flanging effect occurs by cycling over time from an initial delay setting to a second (or final) delay setting.

Final Delay Time Sets the point in milliseconds at which flanging ends behind the original signal.

Stereo Phasing Sets the left and right delays at separate values, measured in degrees. For example, 180° sets the initial delay of the right channel to occur at the same time as the final delay of the left channel. You can set this option to reverse the initial/final delay settings for the left and right channels, creating a circular, psychedelic effect.

Feedback Determines the percentage of the flanged signal that is fed back into the flanger. With no feedback, the effect uses only the original signal. With feedback added, the effect uses a percentage of the affected signal from before the current point of playback.

Modulation Rate Determines how quickly the delay cycles from the initial to final delay times, measured either in cycles per second (Hz) or beats per minute (beats). Small setting adjustments produce widely varying effects.

Mode Provides three ways of flanging:

Inverted Inverts the delayed signal, cancelling out audio periodically instead of reinforcing the signal. If the Original - Expanded mix settings are set at 50/50, the waves cancel out to silence whenever the delay is at zero.

Special Effects Mixes the normal and inverted flanging effects. The delayed signal is added to the effect while the leading signal is subtracted.

Sinusoidal Makes the transition from initial delay to final delay and back follow a sine curve. Otherwise, the transition is linear, and the delays from the initial setting to the final setting are at a constant rate. If Sinusoidal is selected, the signal is at the initial and final delays more often than it is between delays.

Mix Adjusts the mix of original (Dry) and flanged (Wet) signal. You need some of both signals to achieve the characteristic cancellation and reinforcement that occurs during flanging. With Original at 100%, no flanging occurs at all. With Delayed at 100%, the result is a wavering sound, like a bad tape player.

Phaser effect

Similar to flanging, phasing shifts the phase of an audio signal and recombines it with the original, creating psychedelic effects first popularized by musicians of the 1960s. But unlike the Flanger effect, which uses variable delays, the Modulation > Phaser effect sweeps a series of phase-shifting filters to and from an upper frequency. Phasing can dramatically alter the stereo image, creating unearthly sounds.

Stages Specifies the number of phase-shifting filters. A higher setting produces denser phasing effects.

Intensity Determines the amount of phase-shifting applied to the signal.

Depth Determines how far the filters travel below the upper frequency. Larger settings produce a wider tremolo effect; 100% sweeps from the upper frequency to zero Hz.

Mod Rate Modulation rate controls how fast the filters travel to and from the upper frequency. Specify a value in Hz (cycles per second).

Phase Diff Determines the phase difference between stereo channels. Positive values start phase shifts in the left channel, negative values in the right. The maximum values of +180 and -180 degrees produce a complete difference and are sonically identical.

Upper Freq Sets the upper-most frequency from which the filters sweep. To produce the most dramatic results, select a frequency near the middle of the selected audio's range.

Feedback Feeds a percentage of the phaser output back to the input, intensifying the effect. Negative values invert phase before feeding audio back.

Mix Controls the ratio of original to processed audio.

Output Gain Adjusts the output level after processing.

Reverb effects

In a room, sound bounces off the walls, ceiling, and floor on the way to your ears. All these reflected sounds reach your ears so closely together that you don't perceive them as separate echoes, but as a sonic ambience that creates an impression of space. This reflected sound is called *reverberation,* or *reverb* for short. With Adobe Audition, you can use reverb effects to simulate a variety of room environments.

For the most flexible, efficient use of reverb in the Multitrack Editor, add reverb effects to buses, and set reverb output levels to 100% Wet. Then, route tracks to these buses, and use sends to control the ratio of dry to reverberant sound.

Convolution Reverb effect

The Reverb > Convolution Reverb effect reproduces rooms ranging from coat closets to concert halls. Convolution-based reverbs use *impulse* files to simulate acoustic spaces. The results are incredibly realistic and life-like.

Sources of impulse files include audio you've recorded of an ambient space, or impulse collections available online. For best results, impulse files should be uncompressed, 16- or 32-bit files matching the sample rate of the current audio file. Impulse length should be no more than 30 seconds. For sound design, try a variety of source audio to produce unique, convolution-based effects.

Note:

Because Convolution Reverb requires significant processing, you may hear clicks or pops when previewing it on slower systems. These artifacts disappear after you apply the effect.

Impulse Specifies a file that simulates an acoustic space. Click Load to add a custom impulse file in WAV or AIFF format.

Mix Controls the ratio of original to reverberant sound.

Room Size Specifies a percentage of the full room defined by the impulse file. The larger the percentage, the longer the reverb.

Damping LF Reduces low-frequency, bass-heavy components in reverb, avoiding muddiness and producing a clearer, more articulate sound.

Damping HF Reduces high-frequency, transient components in reverb, avoiding harshness and producing a warmer, lusher sound.

Pre-Delay Determines how many milliseconds the reverb takes to build to maximum amplitude. To produce the most natural sound, specify a short pre-delay of 0–10 milliseconds. To produce interesting special effects, specify a long pre-delay of 50 milliseconds or more.

Width Controls the stereo spread. A setting of 0 produces a mono reverb signal.

Gain Boosts or attenuates amplitude after processing.

Full Reverb effect

The Reverb > Full Reverb effect is convolution-based, avoiding ringing, metallic, and other artificial sounding artifacts.

This effect offers some unique options, such as Perception, which simulates room irregularities, Left/Right Location, which places the source off-center, and Room Size and Dimension, which help you realistically simulate rooms that you can customize. To simulate wall surfaces and resonance, you can change the reverb's frequency absorption by using a three-band, parametric EQ in the Coloration section.

When you change reverb settings, this effect creates a temporary *impulse* file, which simulates the acoustic environment you specify. This file can be several megabytes in size, requiring a few seconds to process, so you might have to wait before hearing a preview. The results, however, are incredibly realistic and easy to tailor.

Note:

The Full Reverb effect demands significant processing; for real-time multitrack use, either pre-render this effect or replace it with Studio Reverb. (See Pre-render track effects to improve performance.)

Reverb Settings

Decay Time Specifies how many milliseconds the reverb takes to decay 60 dB. However, depending on the Coloration parameters, certain frequencies may take longer to decay to 60 dB, while other frequencies may decay much faster. Longer values give longer reverb tails, but they also require more processing. The effective limit is about 6000 milliseconds (a 6-second tail), but the actual tail generated is much longer to allow for decaying into the background noise level.

Pre-Delay Time Specifies how many milliseconds reverb takes to build to its maximum amplitude. Generally, reverbs build up quickly, and then decay at a much slower rate. Interesting effects can be heard with extremely long pre-delay times of 400 milliseconds or more.

Diffusion Controls the rate of echo buildup. High diffusion values (above 900 milliseconds) give very smooth reverbs, without distinct echoes. Lower values produce more distinct echoes because the initial echo density is lighter, but the density builds over the life of the reverb tail.

***Tip**: Bouncy echo effects can be obtained by using low Diffusion values and high Perception values. With long reverb tails, using low Diffusion values and somewhat low Perception values gives the effect of a football stadium or similar arena.*

Perception Simulates irregularities in the environment (objects, walls, connecting rooms, and so on). Low values create a smoothly decaying reverb without any frills. Larger values give more distinct echoes (coming from different locations).

***Tip**: If a reverb is too smooth, it may not sound natural. Perception values up to about 40 give simulate typical room variations.*

Room Size Sets the volume of the virtual room, as measured in cubic meters. The larger the room, the longer the reverb. Use this control to create virtual rooms of only a few square meters to giant coliseums.

Dimension Specifies the ratio between the room's width (left to right) and depth (front to back). A sonically appropriate height is calculated and reported as Actual Room Dimensions at the bottom of the dialog box. Generally, rooms with width-to-depth ratios between 0.25 and 4 provide the best sounding reverbs.

Left/Right Location (stereo audio only) Lets you place early reflections off-center. Select Include Direct in the Output Level section to place the original signal in the same location. Very nice effects are possible with singers slightly off center, 5-10% to the left or right.

High Pass Cutoff Prevents the loss of low-frequency (100 Hz or less) sounds, such as bass or drums. These sounds can get phased out when using small rooms if the early reflections mix with the original signal. Specify a frequency above that of the sound you wish to keep. Good settings are generally between 80 Hz and 150 Hz. If the cutoff setting is too high, you may not get a realistic image of the room size.

Set Reverb Based On Room Size Sets Decay and Pre-delay times to match the specified room size, producing a more convincing reverb. If desired, you can then fine-tune the Decay and Pre-Delay times.

Coloration options

To visually adjust Coloration options, drag directly in the graph.

Frequency Specifies the corner frequency for the low and high shelves or the center frequency for the middle band. For example, to increase reverb warmth, lower the high shelf frequency while also reducing its gain.

Gain Boosts or attenuates reverb in different frequency ranges.

***Tip**: To subtly enhance audio, boost reverb frequencies around the natural frequency of a key sonic element. For a singer's voice, for example, boost frequencies from 200 Hz to 800 Hz to enhance resonance in that range.*

Q Sets the width of the middle band. Higher values affect a narrower range of frequencies, and lower values affect a wider range.

***Tip**: For distinct resonance, use values of 10 or higher. To boost or cut a wide range of frequencies, use lower values like 2 or 3.*

Decay Specifies how many milliseconds the reverb decays before the Coloration curve is applied. Values up to 700 work fine. For more colored reverbs, use lower settings (such as 100 to 250).

Output Level options

Dry Controls the level of original signal included with reverb. Use a low level to create a distant sound. Use a high level (near 100%) along with low levels of reverberation and reflections to create a sense of close proximity to the source.

Reverberation Controls the level of the dense layer of reverberant sound. The balance between the dry and reverberant sounds changes perception of distance.

Early Reflections Controls the level of the first echoes to reach the ear, giving a sense of the overall room size. Too high a value can result in an artificial sound, while too low a value can remove audible cues for the room's size. Half the volume of the Dry signal is a good starting point.

Include Direct Slightly phase-shifts the original signal's left and right channels to match the location of early reflections (set by Left/Right Location on the Early Reflections tab).

Sum Inputs Combines the channels of a stereo or surround waveform before processing occurs. Select this option for faster processing, but deselect it for a fuller, richer reverb.

Reverb effect

The Reverb > Reverb effect simulates acoustic spaces with convolution-based processing. It can reproduce acoustic or ambient environments such as a coat closet, a tiled bathroom shower, a concert hall, or a grand amphitheater. The echoes can be spaced so closely together that a signal's reverberated tail decays smoothly over time, creating a warm and natural sound. Alternatively, Pre-Delay Time can be adjusted to give a sense of room size.

Relative to the Reverb effect, the Full Reverb effect provides more options and better audio rendering. For quick adjustments, however, you may prefer the reduced options set of the Reverb effect.

Note:

The Reverb effect demands significant processing; for real-time, multitrack use, either pre-render this effect or replace it with Studio Reverb. (See Pre-render track effects to improve performance.)

Decay Time Sets how many milliseconds it takes for reverb to tail off to infinity (about -96 dB). Use values below 400 for small rooms, values between 400 and 800 for medium-sized rooms, and values above 800 for very large rooms, such as concert halls. For example, enter 3000 milliseconds to create reverb tails for a giant amphitheater.

***Tip**: To simulate rooms that have both echoes and reverb, first use the Echo effect to establish the size of the room, and then use the Reverb effect to make the sound more natural. A Decay Time as little as 300 milliseconds can add perceived spaciousness to dry sound.*

Pre-Delay Time Specifies how many milliseconds reverb takes to build to its maximum amplitude. For a short Decay Time, the Pre-Delay Time time should also be smaller. In general, a value about 10% as long as the Decay Time sounds most realistic. However, you can create interesting effects by using a longer Pre-Delay Time with a shorter Decay Time.

Diffusion Simulates natural absorption, reducing high frequencies as the reverb decays. Faster absorption times simulate rooms full of people, furniture, and carpeting, such as nightclubs and theaters. Slower times (over 1000 milliseconds) simulate empty rooms such as auditoriums, where high frequency reflections are more prevalent.

Perception Changes the characteristics of reflections within a room. Lower values create smoother reverb without as many distinct echoes. Higher values simulate larger rooms, cause more variation in reverb amplitude, and add spaciousness by creating distinct reflections over time.

***Tip**: A Perception setting of 100 and a Decay Time of 2000 milliseconds or more creates interesting canyon effects.*

Dry Sets the percentage of source audio to output. In most cases, 90% works well. To add subtle spaciousness, set the Dry percentage higher; to achieve a special effect, set the Dry percentage lower.

Wet Sets the percentage of reverb to output. To add subtle spaciousness to a track, keep the Wet percentage lower than the Dry percentage. Increase the Wet percentage to simulate greater distance from the audio source.

Sum Inputs Combines the channels of a stereo or surround waveform before processing occurs. Select this option for faster processing, but deselect it for fuller, richer reverb.

Studio Reverb effect

Like the other reverb effects, the Reverb > Studio Reverb effect simulates acoustic spaces. It is faster and less processor-intensive than the other reverb effects, however, because it isn't convolution-based. As a result, you can make real-time changes quickly and effectively in the Multitrack Editor, without pre-rendering effects on a track.

Room Size Sets the room size.

Decay Adjusts the amount of reverberation decay in milliseconds.

Early Reflections Controls the percentage of echoes that first reach the ear, giving a sense of the overall room size. Too high a value can result in an artificial sound, while too low a value can lose the audio cues for the room's size. Half the volume of the original signal is a good starting point.

Stereo Width Controls the spread across the stereo channels. 0% produces a mono reverb signal; 100% produces maximum stereo separation.

High Frequency Cut Specifies the highest frequency at which reverb can occur.

Low Frequency Cut Specifies the lowest frequency at which reverb can occur.

Damping Adjusts the amount of attenuation applied to the high frequencies of the reverb signal over time. Higher percentages create more damping for a warmer reverb tone.

Diffusion Simulates the absorption of the reverberated signal as it is reflected off of surfaces, such as carpeting and drapes. Lower settings create more echoes, while higher settings produce a smoother reverberation with fewer echoes.

Dry Sets the percentage of source audio to output with the effect.

Wet Sets the percentage of reverb to output.

Surround Reverb effect

The Reverb > Surround Reverb effect is primarily intended for 5.1 sources, but it can also provide surround ambience to mono or stereo sources. In the Waveform Editor, you can choose Edit > Convert Sample Type to convert a mono or stereo file to 5.1, and then apply Surround Reverb. In the Multitrack Editor, you can send mono or stereo tracks to a 5.1 bus or master with Surround Reverb.

Input, Center Determines the percentage of the center channel included in the processed signal.

Input, LFE Determines the percentage of the Low Frequency Enhancement channel used to excite reverb for other channels. (The LFE signal itself is not reverberated.)

***note**: The effect always inputs 100% of the Left, Right, and rear surround channels.*

Impulse Specifies a file that simulates an acoustic space. Click Load to add a custom, 6- channel impulse file in WAV or AIFF format.

Room Size Specifies a percentage of the full room defined by the impulse file. The larger the percentage, the longer the reverb.

Damping LF Reduces low-frequency, bass-heavy components in reverb, avoiding muddiness and producing a clearer, more articulate sound.

Damping HF Reduces high-frequency, transient components in reverb, avoiding harshness and producing a warmer, lusher sound.

Pre-Delay Determines how many milliseconds the reverb takes to build to maximum amplitude. To produce the most natural sound, specify a short pre-delay of 0–10 milliseconds. To produce interesting special effects, specify a long pre-delay of 50 milliseconds or more.

Front Width Controls the stereo spread across the front three channels.

A width setting of 0 produces a mono reverb signal.

Surround Width Controls the stereo spread across the rear surround channels (Ls and Rs).

C Wet Level Controls the amount of reverb added to the Center channel. (Because this channel usually contains dialog, reverb should typically be lower.)

L/R Bal. Controls left-right balance for front and rear speakers. 100 outputs reverb to only the left, -100 to only the right.

F/B Bal. Controls front-back balance for left and right speakers. 100 outputs reverb to only the front, -100 to only the back.

Wet/Dry Mix Controls the ratio of original to reverberant sound. A setting of 100 outputs only reverb.

Gain Boosts or attenuates amplitude after processing.

How to use special effects with Audition

Note:

Special effects require mono or stereo audio; they do not support 5.1 surround.

Distortion effect

Use the **Special** > **Distortion** effect to simulate blown car speakers, muffled microphones, or overdriven amplifiers.

Positive and Negative graphs Specify separate distortion curves for positive and negative sample values. The horizontal ruler (x-axis) indicates input level in decibels; the vertical ruler (y-axis) indicates output level. The default diagonal line depicts an undistorted signal, with a one-to-one relationship between input and output values.

Click and drag to create and adjust points on the graphs. Drag points off a graph to remove them.

To copy one graph to another, click the arrow buttons between them.

Reset

Returns a graph to its default, undistorted state.

Curve Smoothing Creates curved transitions between control points, sometimes producing a more natural distortion than the default linear transitions.

Time Smoothing Determines how quickly distortion reacts to changes in input levels. Level measurements are based on low-frequency content, creating softer, more musical distortion.

dB Range Changes the amplitude range of the graphs, limiting distortion to that range.

Linear Scale Changes the amplitude scales of the graphs from logarithmic decibels to normalized values.

Post-filter DC Offset Compensates for any sample offset introduced by distortion processing. To understand this concept, see Correct DC offset. Such offsets can cause audible pops and clicks when edited.

Doppler Shifter (process) effect

The **Special** > **Doppler Shifter (process)** effect creates the increase and decrease in pitch we notice when an object approaches and then passes us, such as when a police car passes with its siren on. When the car comes toward you, the sound reaches your ears as a higher frequency because each sound wave is compressed by the car moving forward. The opposite happens as the car passes by; the waves are stretched out, resulting in a lower-pitched sound.

Note:

*Unlike many graphs in Adobe Audition effects, the **Doppler Shifter** graph is noninteractive: You can't directly manipulate the graph. Instead, the graph changes as you adjust the effect's parameters.*

Path Type Defines which path the sound source appears to take. Depending on the path type, a different set of options is available.

Straight Line options:

- **Starting Distance Away** sets the virtual starting point (in meters) of the effect.
- **Velocity** defines the virtual speed (in meters per second) at which the effect moves.
- **Coming From** sets the virtual direction (in degrees) from where the effect appears to come.
- **Passes In Front By** specifies how far (in meters) the effect seems to pass in front of the listener.
- **Passes On Right By** specifies how far (in meters) the effect seems to pass to the right of the listener.

Circular options:

- **Radius** sets the circular dimensions (in meters) of the effect.
- **Velocity** defines the virtual speed (in meters per second) at which the effect moves.
- **Starting Angle** sets the beginning virtual angle (in degrees) of the effect.
- **Center In Front By** specifies how far (in meters) the sound source is from the front of the listener.
- **Center On Right By** specifies how far (in meters) the sound source is from the right of the listener.

Adjust Volume Based on Distance or Direction Automatically adjusts the effect's volume based on the values specified.

Quality Level Provides six different levels of processing quality. Lower quality levels require less processing time, but higher quality levels generally produce better sounding results.

Guitar Suite effect

The **Special** > **Guitar Suite** effect applies a series of processors that optimize and alter the sound of guitar tracks. The **Compressor** stage reduces dynamic range, producing a tighter sound with greater impact. **Filter**, **Distortion**, and **Box Modeler** stages simulate common effects that guitarists use to create expressive, artistic performances.

Apply the Guitar Suite to vocals, drums, or other audio to create textured effects.

Compressor Reduces dynamic range to maintain consistent amplitude and help guitar tracks stand out in a mix.

Filter Simulates guitar filters ranging from resonators to talk boxes. Choose an option from this menu, and then set options below:

Type Determines which frequencies are filtered. Specify **Lowpass** to filter high frequencies, **Highpass** to filter low frequencies, or **Bandpass** to filter frequencies above and below a center frequency.

Freq Determines the cutoff frequency for **Lowpass** and **Highpass** filtering, or the center frequency for **Bandpass** filtering.

Resonance Feeds back frequencies near the cutoff frequency, adding crispness with low settings and whistling harmonics with high settings.

Distortion Adds a sonic edge often heard in guitar solos. To change the distortion character, choose an option from the **Type** menu.

Amplifier Simulates various amplifier and speaker combinations that guitarists use to create unique tones.

Mix Controls the ratio of original to processed audio.

Mastering effect

Masteringdescribes the complete process of optimizing audio files for a particular medium, such as radio, video, CD, or the web. In Adobe Audition, you can quickly master audio with the **Special** > **Mastering** effect.

Before mastering audio, consider the requirements of the destination medium. If the destination is the web, for example, the file will likely be played over computer speakers that poorly reproduce bass sounds. To compensate, you can boost bass frequencies during the equalization stage of the mastering process.

Equalizer Adjusts the overall tonal balance.

Graph Shows frequency along the horizontal ruler (x-axis) and amplitude along the vertical ruler (y-axis), with the curve representing the amplitude change at specific frequencies. Frequencies in the graph range from lowest to highest in a logarithmic fashion (evenly spaced by octaves).

Drag control points in the graph to visually adjust the settings below.

Low Shelf and High Shelf Enable Activate shelving filters at either end of the frequency spectrum.

Peaking Enable Activates a peaking filter in the center of the frequency spectrum.

Hz Indicates the center frequency of each frequency band.

dB Indicates the level of each frequency band.

Q Controls the width of the affected frequency band. Low Q values (up to 3) affect a larger range of frequencies and are best for overall audio enhancement. High Q values (6–12) affect a very narrow band and are ideal for removing a particular, problematic frequency, like 60-Hz hum.

Reverb Adds ambience. Drag the **Amount** slider to change the ratio of original to reverberant sound.

Exciter Exaggerates high-frequency harmonics, adding crispness and clarity. Mode options include **Retro** for light distortion, **Tape** for bright tone, and **Tube** for quick, dynamic response. Drag the **Amount** slider to adjust the level of processing.

Widener Adjusts the stereo image (disabled for mono audio). Drag the **Width** slider to the left to narrow the image and increase central focus. Drag the slider to the right to expand the image and enhance spatial placement of individual sounds.

Loudness Maximizer Applies a limiter that reduces dynamic range, boosting perceived levels. A setting of 0% reflects original levels; 100% applies maximum limiting.

Output Gain Determines output levels after processing. For example, to compensate for EQ adjustments that reduce overall level, boost the output gain.

Loudness Radar Meter effect

TC Electronic **Loudness Radar** meter plug-in provides you with information about peak, average, and range levels. **Loudness History**, **Momentary Loudness**, **True-peak Level** and flexible descriptors are combined to give you the loudness overview in a single view. The **Radar** sweep view, which provides an excellent view of loudness changes over time, is also available. Choose **Effects** > **Special** > **Loudness Radar** Meter.

TC Loudness Radar

TC Loudness Radar

Settings

Target Loudness Defines the target loudness value

Radar Speed Controls the time of each radar sweep

Radar Resolution Sets the difference in loudness between each concentric circle in the radar view.

Momentary Range Sets the range for the momentary range. EBU +9, indicates the narrow loudness range, used for normal broadcast. EBU +18 is the wide loudness range used for drama and music.

Low Level Below Sets the shift between the green and blue colors on the momentary loudness ring. This indicates that the level may be below the noise floor level.

Loudness Unit Sets the **Loudness** unit to display on the radar.

- LKFS: Loudness unit specified by the International Telecommunication Union (ITU)
- LUFS: Loudness unit specified by the European Broadcasting Union (EBU)
- LU: A unit for loudness relative to target according to EBU and ITU.

Loudness Standard Species the **Loudness** standard.

- **BS.1770-3**: This ITU standard concerns **Broadcast** loudness and **True-peak Level** measurement. This standard uses Leq(K) to measure loudness.
- **Leq(K)**: The loudness is based on a **Leq** measurement employing K-weighting, which is a specific frequency weighting developed by the Communications Research Center (a federal research institute in Ottawa, Canada).

Peak Indicator Sets the maximum **True-Peak** level. If this value is exceeded, the peak indicator is activated.

Vocal Enhancer effect

The **Special** > **Vocal Enhancer** effect quickly improves the quality of voice-over recordings. The **Male** and **Female** modes automatically reduce sibilance and plosives, as well as microphone handling noise such as low rumbles. Those modes also apply microphone modeling and compression to give vocals a characteristic radio sound. The **Music** mode optimizes soundtracks so they better complement a voice-over.

Male Optimizes audio for a man's voice.

Female Optimizes audio for a woman's voice.

Music Applies compression and equalization to music or background audio.

Do more with Adobe Audition

Adobe Audition is a powerful application for recording and mixing audio, podcasting, and sound effect design.

Stereo imagery effects

Some effects let you change the apparent location, or *stereo imagery*, of sounds coming from the speakers.

Center Channel Extractor effect

The Stereo Imagery > Center Channel Extractor effect keeps or removes frequencies that are common to both the left and right channels—in other words, sounds that are panned center. Often voice, bass, and lead instruments are recorded this way. As a result, you can use this effect to bring up the volume of vocals, bass, or kick drum, or you can remove any of them to create a karaoke mix.

Extraction tab Limits extraction to audio that meets certain properties.

Extract Either select audio in the Center, Left, Right, or Surround channel, or select Custom and specify the precise phase degree, pan percentage, and delay time for audio you want to extract or remove. (The Surround option extracts audio that is perfectly out of phase between the left and right channels.)

Frequency Range Sets the range you want to extract or remove. Predefined ranges include Male Voice, Female Voice, Bass, and Full Spectrum. Select Custom to define a frequency range.

Discrimination tab Includes settings that help identify the center channel.

Crossover Bleed Move the slider to the left to increase audio bleed through and make the sound less artificial. Move the slider to the right to further separate center channel material from the mix.

Phase Discrimination In general, higher numbers work better for extracting the center channel, while lower values work better for removing the center channel. Lower values allow more bleed through and may not effectively separate vocals from a mix, but they may be more effective at capturing all the center material. In general, a range from 2 to 7 works well.

Amplitude Discrimination and Amplitude Bandwidth Sum the left and right channels, and create a perfectly out-of-phase third channel that Audition uses to remove similar frequencies. If the amplitude at each frequency is similar, in-phase audio common to both channels is also considered. Lower values for Amplitude Discrimination and Amplitude Bandwidth cut more material from the mix, but may also cut out vocals. Higher values make the extraction depend more on the phase of the material and the less on the channel amplitude. Amplitude Discrimination settings between 0.5 and 10 and Amplitude Bandwidth settings between 1 and 20 work well.

Spectral Decay Rate Keep at 0% for faster processing. Set between 80% and 98% to smooth out background distortions.

Center and Side Channel Levels Specifies how much of the selected signal you want to extract or remove. Move the sliders up to include additional material.

Advanced options Click the triangle to access these settings:

FFT Size Specifies the Fast Fourier Transform size, with low settings improving processing speed, and high settings improving quality. In general, settings between 4096 and 8192 work best.

Overlays Defines the number of FFT windows that overlap. Higher values can produce smoother results or a chorus-like effect, but they take longer to process. Lower values can produce bubbly-sounding background noises. Values of 3 to 9 work well.

Window Width Specifies the percentage of each FFT window. Values of 30% to 100% work well.

Graphic Phase Shifter effect

The **Stereo Imagery** > **Graphic Phase Shifter** effect lets you adjust the phase of a waveform by adding control points to a graph.

Right-click points to access the Edit Point dialog box for precise, numerical control.

Phase shift graph The horizontal ruler (x-axis) measures frequency, while the vertical ruler (y-axis) displays the degree of phase to shift, where zero is no phase shift. You can create simulated stereo by creating a zigzag pattern that gets more extreme at the high end on one channel.

Frequency Scale Sets the values of the horizontal ruler (x-axis) on a linear or logarithmic scale. Select Logarithmic to work at finer detail in the lower frequencies. (The logarithmic scale better reflects the frequency-emphasis of human hearing.) Select Linear to work at finer detail in the higher frequencies.

Range Sets the values of the vertical ruler (y-axis) on a 360° or 180° scale.

Channel Specifies the channel(s) to apply the phase shift to.

Note:

Process a single channel for the best results. If you apply identical phase shift to two stereo channels, the resulting file sounds exactly the same.

FFT Size Specifies the Fast Fourier Transform size. Higher sizes create more precise results, but they take longer to process.

Stereo Expander effect

The Stereo Imagery effect positions and expands the stereo image. Because the Stereo Expander is VST-based, however, you can combine it with other effects in the Mastering Rack and Effects Rack. In Multitrack View, you can also vary the effect over time by using automation lanes.

Choose **Effects > Stereo Imagery > Stereo Expander**, and set the following options:

Center Channel Pan Positions the center of the stereo image anywhere from hard left (-100%) to hard right (100%).

Stereo Expand Expands the stereo image from Narrow/Normal (0) to Wide (300). Narrow/Normal reflects the original, unprocessed audio.

Time and pitch manipulation effects

Automatic Pitch Correction effect

The **Automatic Pitch Correction** effect is available in both the **Waveform** and **Multitrack** editors. In the latter, its parameters can be automated over time using keyframes and external control surfaces.

Choose **Effects** > **Time and Pitch** > **Automatic Pitch Correction**, and set the following options:

Scale Specifies the scale type that best suits the material: **Major**, **Minor**, or **Chromatic**. Major or Minor correct notes to the specific key of the music. Chromatic corrects to the nearest note regardless of key.

Key Sets the intended key for corrected material. This option is available only if **Scale** is set to **Major** or **Minor** (because the **Chromatic** scale includes all 12 tones and isn't key-specific).

The combination of scale and key determines the key signature.

Attack Controls how quickly Adobe Audition corrects the pitch toward the scale tone. Faster settings are usually best for notes of short duration, such as a fast, staccato passage. An extremely fast attack can achieve a robotic quality, however. Slower settings result in more natural-sounding correction on longer sustaining notes, such as a vocal line where the singer holds notes and adds vibrato. Because source material can change throughout a musical performance, you can get the best results by separately correcting short musical phrases.

Sensitivity Defines the threshold beyond which notes aren't corrected. **Sensitivity** is measured in cents, and there are 100 cents per semitone. For example, a **Sensitivity** value of 50 cents means a note must be within 50 cents (half a semitone) of the target scale tone before it is corrected automatically.

Reference Channel Choose a source channel in which pitch changes are most clear. The effect analyzes only the channel you choose, but applies the pitch correction equally to all channels.

FFT Size Sets the Fast Fourier Transform size of each pieces of data that the effect processes. In general, use smaller values for correcting higher frequencies. For voices, a setting of 2048 or 4096 sounds most natural. For short, staccato notes or percussive audio try a setting of 1024.

Calibration Specifies the tuning standard for the source audio. In Western music, the standard is A4 at 440 Hz. Source audio, however, may have been recorded using a different standard, so you can specify A4 values from 410 to 470 Hz.

Correction meter When you preview audio, displays the amount of correction for flat and sharp tones.

Manual Pitch Correction effect

The **Manual Pitch Correction** effect lets you visually adjust pitch with the **Spectral Pitch Display**. The **Spectral Pitch Display** shows the fundamental pitch as a bright blue line, and overtones in yellow to red hues. Corrected pitch appears as a bright green line.

*You can visually monitor pitch at any time, without using the **Manual Pitch Correction** effect. Simply click the **Spectral Pitch Display** icon in the options bar. To customize resolution, decibel range, and gridlines, adjust Pitch Display settings in the Spectral Displays preferences.*

1. Choose **Effects** > **Time and Pitch** > **Manual Pitch Correction**.
2. In the **Manual Pitch Correction** window, set the following options:

   Reference Channel Choose a source channel in which pitch changes are most clear. The effect analyzes only the channel you choose, but applies the pitch correction equally to all channels.

   Spline Curves Create smoother transitions when using envelope keyframes to apply different pitch correction over time.

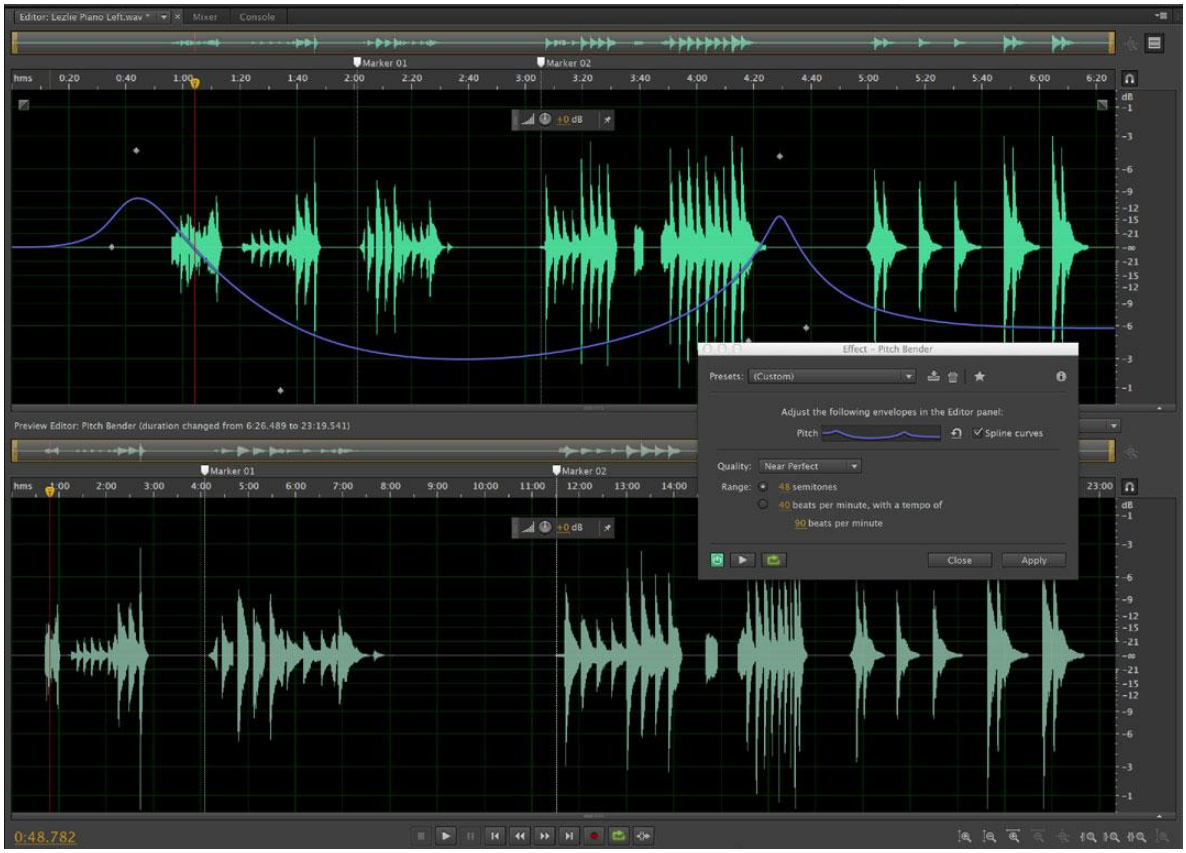
   Pitch Curve Resolution Sets the Fast Fourier Transform size of each piece of data that the effect processes. In general, use smaller values for correcting higher frequencies. For voices, a setting of 2048 or 4096 sounds most natural, and a setting of 1024 creates robotic effects.
3. In the **Editor** panel, do either of the following:
   - To change pitch uniformly, drag the **Adjust Pitch** knob in the heads-up display.
   - To change pitch over time, add keyframes to the yellow envelope line in the center of the waveform display.

To zoom in on specific pitch ranges, right-click and drag in the vertical ruler to the right of the Spectral Pitch Display. To reset the zoom level or customize the displayed scale, right-click the ruler and select options from the pop-up menu.

Pitch bender effect

Use the Pitch Bender effect to change the tempo over time to vary the pitch. The effect now uses a keyframe edit envelope laid across the entire waveform, similar to the Fade and Gain Envelope effects.

Choose **Effects** > **Time and Pitch** > **Pitch Bender**, and set the following options:



Pitch In the Editor panel, click the blue envelope line to add keyframes, and drag them up or down to change amplitude. To quickly select, reposition, or delete multiple keyframes, see Adjust automation with keyframes.

*Select the **Spline Curves** option to create smoother, curved transitions between keyframes, rather than linear transitions. See About spline curves for graphs.*

Quality Controls the quality level. Higher quality levels produce the best sound, but they take longer to process. Lower quality levels produce more unwanted harmonic distortion, but they take less time to process. Usually, you won't notice

harmonic distortion for levels from Very Good and higher. Aliasing still occurs, however, when you shift the pitch up, but the higher quality levels greatly reduce the distortion when you shift the pitch down.

Range Sets the scale of the vertical ruler (y-axis) as semitones (there are 12 semitones to an octave) or as beats per minute. For a range in semitones, the pitch changes logarithmically, and you can specify the number of semitones to shift up or down. For a range in beats per minute, the pitch changes linearly, and you must specify both a range and a base tempo. You can specify the exact tempo of a selection to change to different rates, but this isn't required.

Pitch shifter

The Pitch Shifter effect changes the musical pitch. It's a real-time effect which can be combined with other effects in the mastering rack or the effects rack. In the Multitrack View, you can also vary pitch over time by using automation lanes.

Choose Effects > Time and Pitch > Pitch Shifter effect, and set the following options:

Pitch Transpose Contains options that adjust pitch:

- **Semi-Tones** Transposes pitch in semi-tone increments, which equalmusical half-notes (for example, the note C# is one semi-tone higher than C). A setting of 0 reflects the original pitch; +12 semi-tones is an octave higher; -12 semi-tones is an octave lower.
- **Cents** Adjusts pitch in fractions of semi-tones. Possible values range from -100 (one semi-tone lower) to +100 (one semi-tone higher).
- **Ratio** Determines the relationship between shifted and original frequency. Possible values range from 0.5 (an octave lower) to 2.0 (an octave higher).

Precision Determines sound quality, with theHigh setting taking longest to process. Use the Low setting for 8-bit or low-quality audio, and use the High setting for professionally recorded audio.

To quickly determine which Precision setting to use, process a small selected range at each setting until you find the best balance of quality and processing time.

Pitch Settings Control how audio is processed:

- **Splicing Frequency** Determines the size of each chunk of audio data. (The Pitch Shifter effect divides audio into very small chunks for processing.) The higher the value, the more precise the placement of stretched audio over time. However, artifacts become more noticeable as values go up. At higher Precision settings, a lower Splicing Frequency may add stutter or echo. If the frequency is too high, sound becomes tinny and voices have a tunnel-like quality.
- **Overlapping** Determines how much each chunk of audio data overlaps with the previous and next ones. If stretching produces a chorus effect, lower the Overlapping percentage. If doing so produces a choppy sound, adjust the percentage to strike a balance between choppiness and chorusing. Values range from 0 to 50%.
- **Use Appropriate Default Settings** Applies good default values for Splicing Frequency and Overlapping.

Stretch and Pitch effect (Waveform Editor only)

The Time And Pitch > Stretch And Pitch effect lets you change the pitch of an audio signal, the tempo, or both. For example, you can use the effect to transpose a song to a higher key without changing the tempo, or you can use it to slow down a spoken passage without changing the pitch.

Note:

This effect requires offline processing. While it is open, you cannot edit the waveform, adjust selections, or move the current-time indicator.

Algorithm Choose IZotope Radius to simultaneously stretch audio and shift pitch, or Audition to change stretch or pitch settings over time. The iZotope Radius algorithm requires longer processing but introduces fewer artifacts.

Precision Higher settings produce better quality but require more processing time.

New Duration Indicates how long the audio will be after time-stretching. You can either adjust the New Duration value directly, or indirectly by changing the Stretch percentage.

If you commonly stretch files to a certain duration, click the Favorite icon *to save that setting for future use. To apply a favorite to multiple files, see Batch process files.*

Lock Stretch Settings to New Duration Overrides custom or preset Stretch settings, instead calculating them from duration adjustments.

Select the option above to quickly make radio spots 30 or 60 seconds long.

Stretch Shortens or extends processed audio relative to existing audio. For example, to shrink audio to half its current duration, specify a Stretch value of 50%.

Pitch Shift Tonally shifts audio up or down. Each semitone equals one half-step on a keyboard.

Final Stretch or Pitch Shift (Audition algorithm) Changes the initial Stretch or Pitch Shift setting over time, reaching the final setting at the last selected audio sample.

Lock Stretch and Pitch Shift (IZotope algorithm) Stretches audio to reflect pitch changes, or vice versa.

Lock Initial Strech and Pitch Shift (Audition algorithm) Stretches audio to reflect pitch changes, or vice versa. Final Stretch or Pitch Shift settings are unaffected.

Advanced settings (IZotope Radius algorithm) Click the triangle to access these options:

Solo Instrument Or Voice More quickly processes a solo performance.

Preserve Speech Characteristics Maintains realism in speech.

Formant Shift Determines how formants adjust to pitch shifts. The default value of zero adjusts formants together with pitch shifts, maintaining timbre and realism. Values above zero produce higher timbres (making a male voice sound female, for example). Values below zero do the reverse.

Pitch Coherence Maintains timbre of solo instruments or vocals. Higher values reduce phasing artifacts but introduce more pitch modulation.

Advanced settings (Audition algorithm) Click the triangle to access these options:

Splicing Frequency Determines how big each chunk of audio data is when you preserve pitch or tempo while stretching a waveform. The higher the value, the more precise the placement of stretched audio over time. However, artifacts are more noticeable as rates go up; sound can become tinny or have a tunnel-like quality. With higher Precision settings, lower splicing frequencies may add stutter or echo.

Overlapping Determines how much each chunk of audio data overlaps with the previous and next ones. If stretching produces a chorus effect, lower the Overlapping percentage, without going so low that you produce a choppy sound. Overlapping can be as high as 400%, but you should use this value only for very high speed increases (200% or more).

Choose Appropriate Defaults Applies good default values for Splicing Frequency and Overlapping. This option is good for preserving pitch or tempo.

Constant Vowels Preserves the sound of vowels in stretched vocals. This option requires substantial processing; try it on a small selection before applying it to a larger one.

Generate tones and noise

Generate Noise

The Noise command lets you generate random noise in a variety of colors. (Traditionally, color is used to describethe spectral composition of noise. Each color has its own characteristics.) Generating noise is useful for creating soothing sounds like waterfalls (perfect for use with the Binaural Auto-Panner function of Adobe Audition) and for generating signals that can be used to check out the frequency response of a speaker, microphone, or other audiosystem component.

1. Place the cursor where you want to insert the noise. Or, if you want to replace part of the existing waveform, select the desired range of audio data.
2. Choose Effects > Generate > Noise.
3. Set options as desired, and click OK.

Color Specifies a color for the noise:

- **Brown noise** - Has a spectral frequency of 1/f2, which means, in layman's terms, that the noise has much more low-frequency content. Its sounds are thunder- and waterfall-like. Brown noise is so called because, when viewed, the wave follows a Brownian motion curve. That is, the next sample in the waveform is equal to the previous sample, plus a small random amount. When graphed, this waveform looks like a mountain range.
- **Pink noise** - Has a spectral frequency of 1/f and is found mostly in nature. It is the most natural sounding of the noises. By equalizing the sounds, you can generate rainfall, waterfalls, wind, rushing river, and other natural sounds. Pink noise is exactly between brown and white noise (hence, some people used to call it tan noise). It is neither random nor predictable; it is fractal-like when viewed. When zoomed in, the pattern looks identical to when zoomed out, except at a lower amplitude.
- **White noise** - Has a spectral frequency of 1, meaning that equal proportions of all frequencies are present. Because the human ear is more susceptible to high frequencies, white noise sounds very hissy. Adobe Audition generates white noise by choosing random values for each sample.

Style Specifies a style for the noise:

- **Spatial Stereo** - Generates noise by using three unique noise sources and spatially encoding them to seem as if one comes from the left, one from the center, and one from the right. When you listen to the result with stereo headphones, your mind perceives sound coming from all around. To specify the distance from center of the left and right noise sources, enter a delay value in microseconds. About 900 to 1000 microseconds correspond to the maximum delay perceivable. A delay of zero is identical to monaural noise, where left and right channels are the same.
- **Independent Channels** - Generates noise by using two unique noise sources, one for each channel. The left channel's noise is completely independent of the right channel's noise.
- **Mono** - Generates noise by using a single noise source, with the left and right channels set equally to that source.
- **Inverse** - Generates noise by using a single noise source (similar to the Mono option). However, the left channel's noise is exactly inverse of the right channel's noise. When you listen to the result with stereo headphones, your mind perceives sound coming from within your head instead of from somewhere externally.

Intensity Specifies the intensity of the noise on a scale of 2 to 40. At higher intensities, the noise becomes more erratic and sounds harsher and louder.

DC Offset Adds a constant DC (Direct Current) amplitude to the tone, centering the waveform by shifting it up or down by the specified percentage.

Duration Determines the number of seconds of noise that Adobe Audition generates.

Generate Tones

Choose Effects > Generate > Tones to create a simple waveform using several amplitude- and frequency-related settings. Generated tones are great starting points for sound effects.

Sweep Frequencies Transitions tones from settings on Start tab to those on End tab. (Waveform remains constant.)

Base frequency Specifies the main frequency to be used for generating tones.

Modulation Depth Modulates pitch of the base frequency over a user-defined range. For example, a 100 Hz setting modulates the original frequency from 50 Hz below to 50 Hz above.

Modulation Rate Specifies how many times per second the frequency modulates, producing a warbling, vibrato effect.

Waveform Choose one of the following:

- Sine and Inverse Sine produce the fundamental frequency. A Type setting of 1.00 produces pure tones; lower settings produce more square waves; higher settings more triangular.
- Triangle/Sawtooth produces a true triangle waveform with only odd harmonics at a Type setting of 50%. Settings above and below that percentage produce Sawtooth waveforms with both odd and even harmonics.
- Square produces only odd harmonics. A true square wave occurs with a Type setting of 50%. Settings above and below adjust the duty cycle of the waveform between a flat peak at 100% and a flat trough at 0%. (The outer extremes produce only an audible click.)

Frequency Components Adds up to five overtones to the fundamental frequency (Base Frequency). Below the sliders, enter either a specific frequency for each overtone, or specify a multiplier of the fundamental. Then, use the Amplitude sliders to mix overtones in proportion to one another.

Volume Specifies the overall output of the effect.

Duration Specifies the length of the generated tone in the format currently specified for the time ruler. Right-click the numbers to choose a separate time format for Generate Tones.

To access the following options, click Advanced:

Start Phase At Specifies the starting location in the waveform cycle. If set to 0°, waves will start at the zero-crossing point. If set to 90°, the wave will start at full amplitude (generating a noticeable click).

2nd Channel Phase Difference Adjusts the relative phase of stereo channels. A value of zero places the channels completely in phase and 180 places them completely out of phase.

Change Speed Dynamically changes the relative phase between the two channels of a stereo audio file over time. For example, if you enter 1 Hz, the phase difference will cycle through 360° each second.

DC Offset Adds a Direct Current offset, shifting the center of the waveform up or down by the specified percentage. Use this adjustment to calibrate audio hardware or simulate hardware-induced offset.

With Selected Audio Choose from the following:

- Replace: Swaps existing audio with generated tones.
- Modulate: Ring modulates, or multiplies, existing audio with current tone settings. This option is great for adding special effects.
- Demodulate: Ring demodulates existing audio with current tone settings. Use this option to create unique effects or revert audio previously processed with the Modulate option to its original state.
- Overlap: Mixes the generated tones with existing audio.

Do more with Adobe Audition

Adobe Audition is a powerful application for recording and mixing audio, podcasting, and sound effect design.

Chapter 7: Mixing multitrack sessions

Creating remix

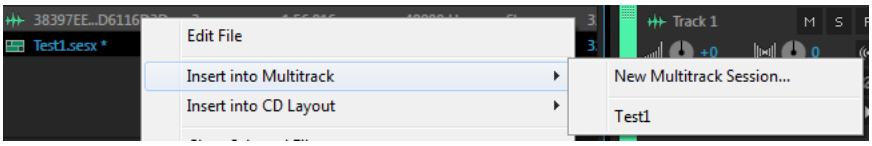
Overview

In Audition CC, you can create remixes of music files from a collection. For example, you can use this option for songs that change from beginning to end, and minimize abrupt changes in a song's dynamics or tempo. You can recompose any piece of music in your collection to fit the video or project duration.

Remix uses a combination of beat detection, content analysis, and spectral source separation technology, to identify hundreds of transition points in your song. It then rearranges the passages to create a composition.

Creating a remix

1 Insert a clip into a multitrack session or create a multitrack session.

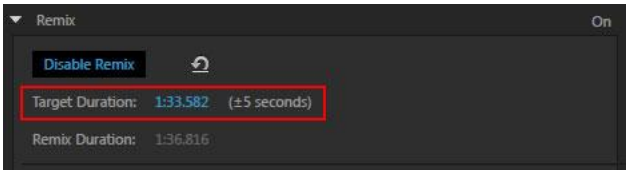


2 Enable the clip for remix in one of the following ways:

- Choose **Clip > Remix > Enable Remix** in the menu bar.
- Right-click the clip and choose **Remix > Enable Remix**.
- Select the clip and in the Remix group of the Properties panel, click **Enable Remix**.

3 Resize the clip to the target duration in one of the following ways:

- Grab the zigzag handles at the top edge of the clip and drag them to the length you require.
- Adjust the duration in **Target Duration** in the **Remix** group.



4 Expand the **Advanced** section in the **Properties** panel to adjust the remix. The available options are:

   a **Stretch to exact duration:** This option enables time-stretch on the remixed clip and adjusts it so that it precisely matches the target duration.

   b **Edit Length:**

      c **Short:** Short results in shorter segments, but in more transitions. This option is suited for songs that changes from beginning to end to minimize abrupt changes in dynamics or tempo.

      d **Long:** Long looks for the longest passages and smallest number of segments to minimize transitions.

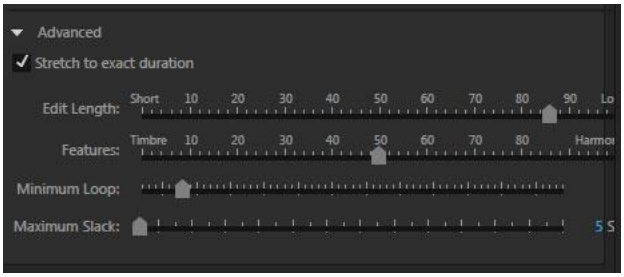
   e **Features:**

      f **Timbre:** Timbre emphasizes tonal qualities in identifying transition points.

      g **Harmonic:** Harmonic emphasizes melodic and harmonic components in transition points.

   h **Minimum Loop:** Set the minimum acceptable length, in beats, to use in a segment.

   i **Maximum Slack:** Set the maximum difference in duration from your target duration.



5 To disable remix and restore the clip to its original version, choose **Clip > Remix > Revert Remix**.

Multitrack Editor overview

About multitrack sessions

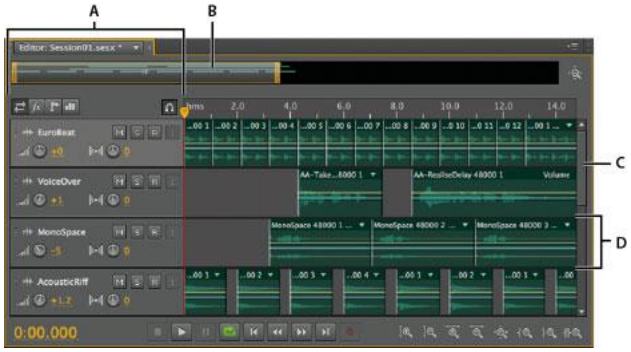
In the Multitrack Editor, you can mix together multiple audio tracks to create layered soundtracks and elaborate musical compositions. You can record and mix unlimited tracks, and each track can contain as many clips as you need—the only limits are hard disk space and processing power. When you're happy with a mix, you can export a mixdown file for use on CD, the web, and more.

The Multitrack Editor is an extremely flexible, real-time editing environment, so you can change settings during playback and immediately hear the results. While listening to a session, for example, you can adjust track volume to properly blend tracks together. Any changes you make are impermanent, or *nondestructive*. If a mix doesn't sound good next week, or even next year, you can simply remix the original source files, freely applying and removing effects to create different sonic textures.

Adobe Audition saves information about source files and mix settings in session (.sesx) files. Session files are relatively small because they contain only pathnames to source files and references to mix parameters (such as volume, pan, and effect settings). To more easily manage session files, save them in a unique folder with the source files they reference. If you later need to move the session to another computer, you can simply move the unique session folder.

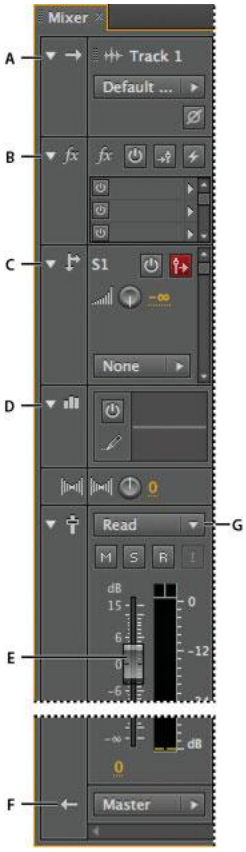
Editing multitrack sessions in the Editor panel and Mixer

In the Multitrack Editor, the Editor panel provides several elements that help you mix and edit sessions. In the track controls on the left, you adjust track-specific settings, such as volume and pan. In the timeline on the right, you edit the clips and automation envelopes in each track.



Editor panel in Multitrack Editor
***A** Track controls **B** Zoom navigator **C** Vertical scroll bar **D** Track*

The Mixer (Window > Mixer) provides an alternative view of a session, revealing many more tracks and controls simultaneously, without showing clips. The Mixer is ideal for mixing large sessions with many tracks.



Controls in the Mixer:
***A** Inputs **B** Effects **C** Sends **D** Equalization **E** Volume **F** Outputs*

Select ranges in the Multitrack Editor

Simultaneously selecting a range and clips in the Editor panel

1 In the toolbar, select the Time Selection tool .

2 In the Editor panel, do one of the following:

- To select only a range, click an empty area of the track display, and drag left or right.
- To select a range and clips, click the center of a clip, and drag a marquee.

Customize start offset and time display for multitrack sessions

1 In the Editor panel, click an empty area of the track display to ensure that no clips are selected.

2 Choose Window > Properties.

3 In the Properties panel, adjust the following options:

Start Time Sets a start-time offset, helping you match audio in Adobe Audition to the time displayed in video applications.

Advanced settings To customize Time Display settings for the active session, set the Time Format and Custom Frame Rate settings. For details, see Change the time display format.

Basic multitrack controls

Understanding video, audio, bus, and master tracks

Multitrack sessions can include four different types of tracks:

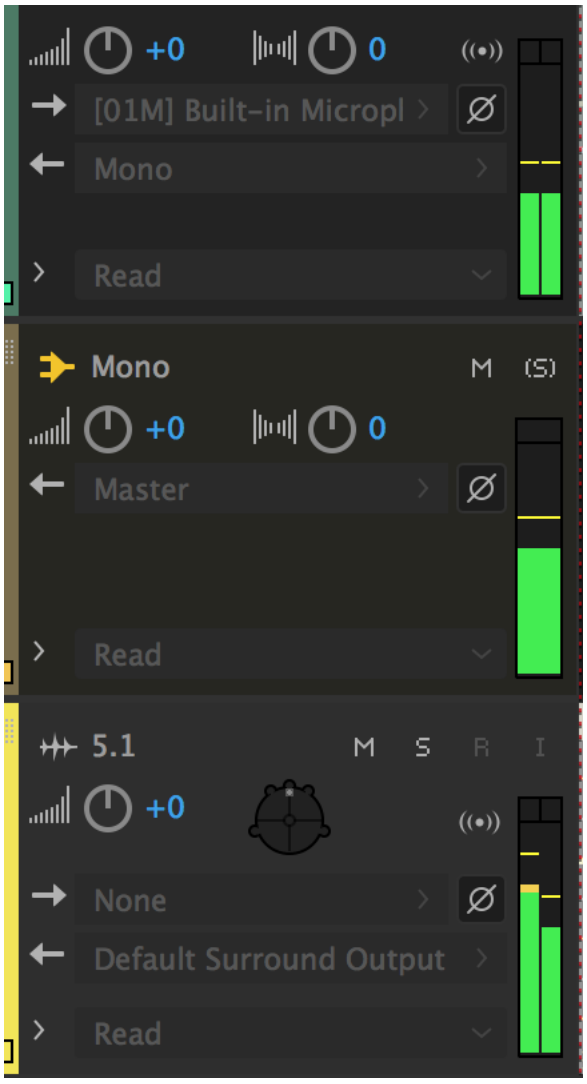
- Video tracks contain an imported video clip. A session can include one video track and clip at a time. You can watch a preview in the Video panel (Window > Video).
- Audio tracks contain either imported audio or clips recorded in the current session. These tracks offer the widest range of controls. You can specify inputs and outputs, apply effects and equalization, route audio to sends and buses, and automate mixes.
- Bus tracks let you combine the outputs of several audio tracks or sends and control them collectively.
- The Master track , which is the last in each session, lets you easily combine the outputs of multiple tracks and buses. You can also control them with a single fader.

For more information, see Routing audio to buses, sends, and the Master track.

Choose between mono, stereo, and 5.1 tracks

Multitrack sessions support an unlimited number of mono, stereo, and 5.1 audio and bus tracks. When adding tracks, choose a channel configuration based on the configuration of the Master track:

- For mono masters, add mono audio and bus tracks.
- For stereo masters, add stereo audio and bus tracks. Except for audio tracks with mono sources (like a single microphone).
- For 5.1 masters, add stereo audio tracks and 5.1 bus tracks (to maintain proper gain staging). If you have any 5.1 audio clips, however, add 5.1 audio tracks to contain them.



To identify track channels, note the meters in the Editor or Mixer panel:
***A** Mono **B** Stereo **C** 5.1*

For information about converting a waveform between surround, stereo, and mono, see the following:

- Convert a waveform between surround, stereo, and mono.

Add or delete tracks

Note:

A multitrack session supports only one video track, which Adobe Audition always inserts at the top of the Editor panel.

Add track

- To add a track, select the track you want to precede it, and then choose **Multitrack > Track > Add [type of] Track**.

Add multiple tracks

- To add multiple tracks, choose **Multitrack > Add Tracks**. You can enter the number of tracks you want to add and the channel layout. Audition also lets you add Bus tracks. You can also choose the channel layout for Bus tracks.

Delete track

- To delete a track, select the track and choose **Multitrack > Track > Delete Selected Track.**

Delete empty tracks

To delete all empty tracks at once, select **Multitrack > Delete empty tracks**.

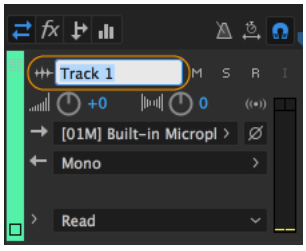
For more information, see the following:

- Insert an audio file into a multitrack session
- Insert a video file into a multitrack session

Name or move tracks

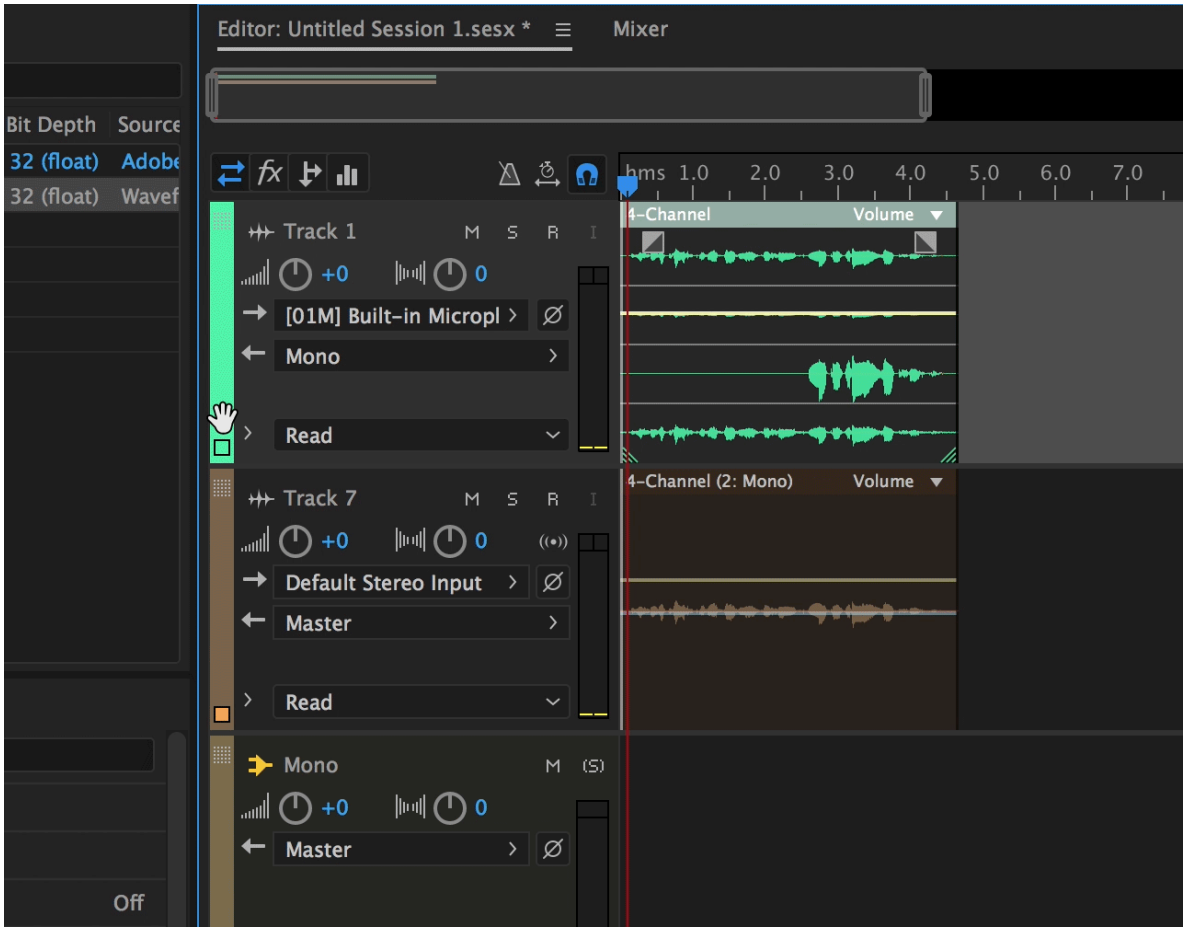
You can name tracks to better identify them, or move them to display related tracks together.

- In the Editor panel or Mixer, type in the name text box.



Name text box in the Editor panel

- Position the pointer to the left of the track name, and then drag either up or down in the **Editor** panel. You can also drag them right or left in the **Mixer**.

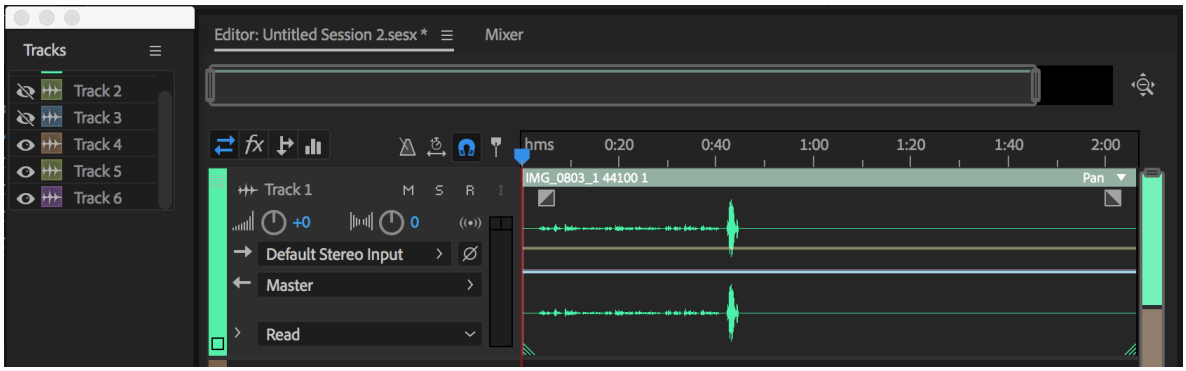


Moving a track in the Editor panel

Show or hide tracks

While working on the multitrack editor, you can use the **Tracks** panel to customize tracks. You can use the panel to choose the tracks you want to see or hide while working on a project. To open the **Tracks** panel, follow this step:

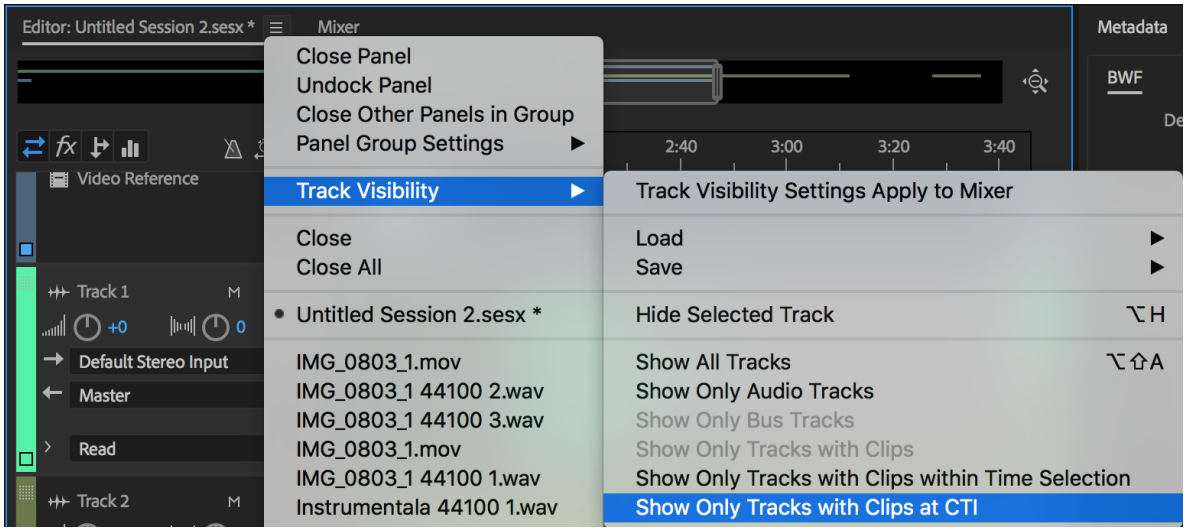
- Select **Windows** > **Tracks**.



Tracks panel

Use **Track Visibility** to see the available presets for tracks. To open **Track Visibility** options, do one of the following:

- To open **Track Visibility** from Tracks panel, select the **Tracks** pop-up window> **Track Visibility.**
- To open **Track Visibility** from Editor panel, select the **Editor** pop-up window > **Track Visibility.**
- To open Track Visibility from the main menu, select **Multitrack** > **Track Visibility**.



Track visibility options

To save custom presets, select **Track Visibility** > **Save** > **Preset**. You can save up to nine presets.

To hide tracks, use the eye icon. indicates that the track is visible, and indicates that the track is hidden.

Vertically zoom tracks

When you use the vertical Zoom options in the lower right of the Editor panel, all tracks zoom simultaneously. If a session contains many tracks, it is advised to zoom them individually.

- In the track controls, drag the top or bottom border of the track up or down.

To quickly zoom all tracks, roll the mouse wheel over the track controls. To horizontally resize all track controls, drag the right border.

For more techniques, see Zoom audio in the Editor panel.

Mute and solo tracks

You can mute solo tracks to hear them separately from the rest of a mix. Conversely, you can mute tracks to silence them in a mix.

- To mute a track, click its Mute button in the Editor panel or Mixer.
- To solo a track, click its Solo button in the Editor panel or Mixer. To automatically remove other tracks from Solo mode, Ctrl-click (Windows) or Command-click (Mac OS).

***Tip**: To remove other tracks from Solo mode by default, select Track Solo: Exclusive in the Multitrack section of the Preferences dialog box. (Regardless of this setting, when you solo a bus, assigned tracks are always placed in Solo mode.)*

Apply an identical setting to all tracks

To increase your efficiency, you can quickly apply several settings to an entire session.

? Hold down Ctrl+Shift (Windows) or Command+Shift (Mac OS). Then select an Input, Output, Mute, Solo, Arm For Record, or Monitor Input setting for any track.

For more information, see the following:

- Assign audio inputs and outputs to tracks
- Record audio clips in the Multitrack Editor

Set track output volume

? Do any of the following:

- In the **Editor** panel, drag the Volume knob . To change settings in large increments, hold down Shift. To change settings in small increments, hold down Ctrl (Windows) or Command (Mac OS)
- In the **Mixer,** drag the track fader, or click above or below to move incrementally to the next tick mark. To move to a specific point, Alt-click (Windows) or Option-click (Mac OS) above or below the fader.

To return knobs and faders to zero (unity gain), Alt-click (Windows) or Option-click (Mac OS) directly on them.

Pan tracks in stereo

- Drag the Pan knob in the Editor panel or Mixer.

To change settings in large increments, hold down Shift. To change settings in small increments, hold down Ctrl (Windows) or Command (MacOS)Tip: By default, the Multitrack Editor pans using an equal power method, which maintains consistent perceived loudness by slightly boosting one channel over the other. To change the boost amount, or switch to logarithmic panning (which simply attenuates one channel), change Panning Mode in the Multitrack preferences.

Duplicate tracks

To perfectly copy all clips, effects, equalization, and envelopes in a track, duplicate it. Duplicate tracks provide a great starting point for new adjustments, helping you compare different processing and automation settings.

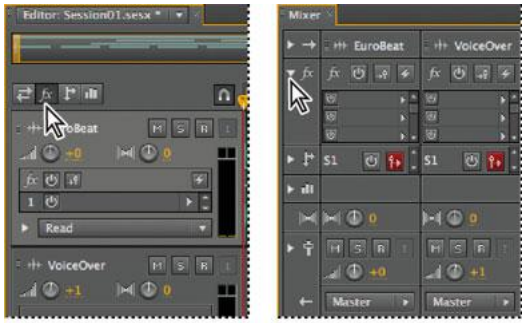
1 In the Editor panel or Mixer, select a track.

2 Choose Multitrack > Track > Duplicate Selected Track.

Multitrack routing and EQ controls

Show or hide track routing and EQ controls

Although the wide variety of routing and EQ controls may seem intimidating at first, the controls for each track are identical, so once you've learned one, you've learned them all.

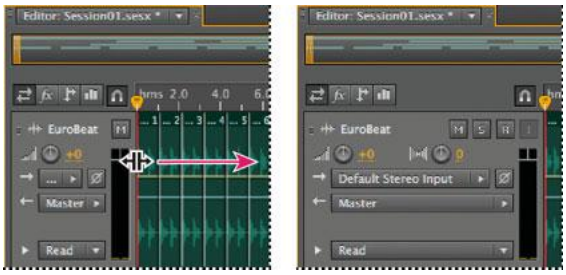


Showing and hiding sets of audio track controls
***A** Editor panel **B** Mixer*

? Do either of the following:

- On the left side of the Mixer, click the Show/Hide triangle for one or more sets of controls.
- In the upper left corner of the Editor panel, click the button for Inputs/Outputs, Effects, Sends, or EQ.

  Tip: In the Editor panel, drag the right or bottom border of the track controls to show more or less detail.



Dragging the right border of the track controls to reveal more or less detail

For more information, see Vertically zoom tracks.

Assign audio inputs and outputs to tracks

? In the Inputs/Outputs area of the Editor panel or Mixer, do the following:

- From the Input menu, choose a hardware input.
- From the Output menu, choose a bus, the Master track, or a hardware output.

The list of available hardware ports is determined by settings in the Audio Hardware preferences. (See Configure audio inputs and outputs.)

For more information, see the following:

- Record audio clips in the Multitrack Editor
- Set track output volume

Invert the polarity of an input

If a pair of stereo inputs such as overhead drum microphones are out-of-phase, you'll hear duller sound and a narrower stereo image. To correct the problem, invert the polarity of one the inputs.

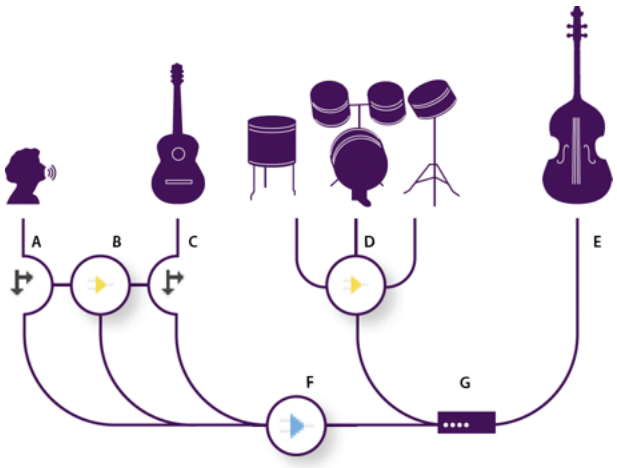
? In the Input Controls area of the Mixer, click the Polarity Reverse button .

To understand audio phase, see How sound waves interact.

Routing audio to buses, sends, and the Master track

Buses, sends, and the Master track let you route multiple track outputs to one set of controls. With these combined controls, you can efficiently organize and mix a session.

To save the output of an audio, bus, or Master track to a file, see Export multitrack mixdown files.

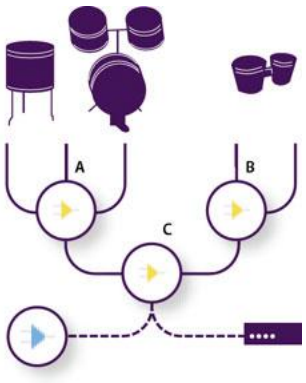


An example of audio routing for tracks
***A** Vocal **B** Reverb bus receiving vocal and guitar sends **C** Guitar **D** Drums bus combining drum outputs **E** Bass outputting direct to hardware **F** Master track **G** Hardware outputs*

Understanding bus tracks

With bus tracks, you can combine the outputs of several audio tracks or sends and control them collectively. For example, to control the volume of multiple drum tracks with a single fader, output all the tracks to one bus. Or, to optimize system performance, apply a single reverb effect to a bus track, and then output sends from multiple tracks to that bus. (Individually applying the same reverb to multiple tracks would inefficiently use CPU resources.)

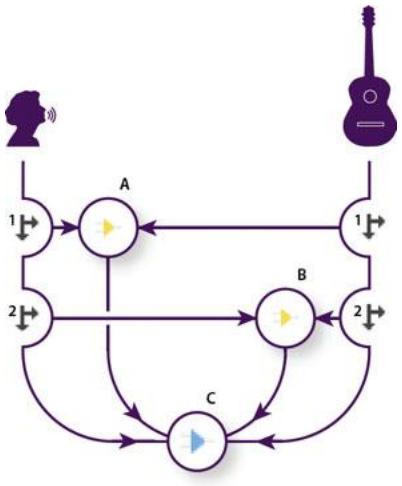
Although bus tracks lack a hardware input, they have all the other features of audio tracks. You can apply effects and equalization and automate your changes over time. For most mixes, you'll output buses to hardware ports or the Master track. If you need to combine buses, however, you can even output them to other buses.



An example of bus routing:
***A** Drum kit bus **B** Hand drum bus **C** Combined drums bus outputting to either the Master track or hardware*

Understanding sends

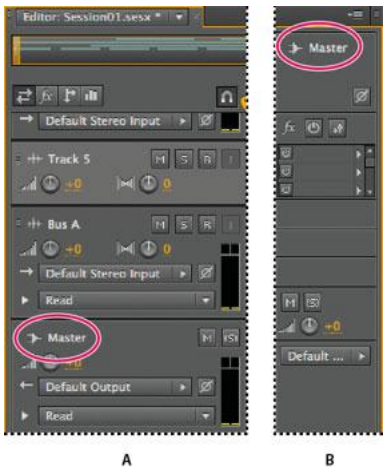
Sends let you route audio from a track to multiple buses, creating tremendous signal-routing flexibility. Each track provides up to 16 sends, which you configure independently from the track output. For example, you can output an unprocessed track directly to a hardware port, but output Send 1 to a reverb bus and Send 2 to a headphone bus. (A headphone bus lets performers hear a unique mix during recording. Drummers, for example, may prefer a louder bass track.)



Sending tracks to multiple buses
***A** Send 1 outputs to delay bus **B** Send 2 outputs to reverb bus **C** Master track combines vocal, guitar, delay, and reverb outputs*

Understanding the Master track

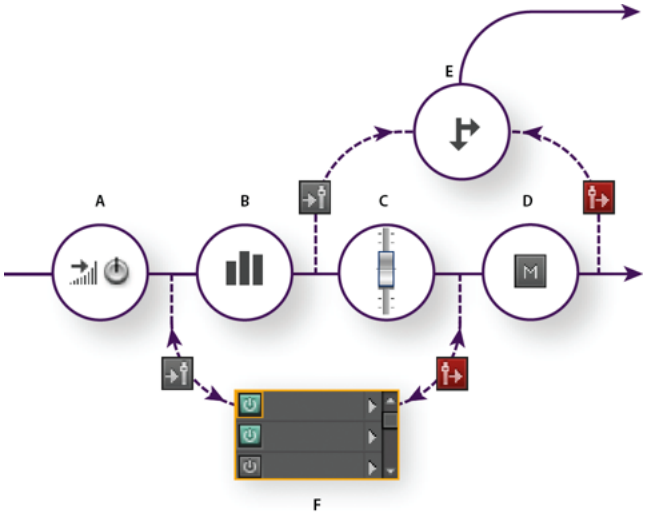
A session always contains one Master track, so you can easily combine the outputs of multiple tracks and buses and control them with a single fader. Because the Master track exists at the very end of the signal path, it offers fewer routing options than audio and bus tracks. The Master track can't directly connect to audio inputs, or output to sends or buses; it can only output directly to hardware ports.



The Master track always exists at the end of the signal path.
***A** Editor panel **B** Mixer*

Set up a send

When you set up a send, you determine the volume and stereo pan it outputs to an assigned bus. You also place the send either pre- or post-fader. Pre-fader sends aren't affected by track volume; post-fader sends are. (For example, if you output a pre-fader send to a reverb bus, the reverb continues after you fade out dry audio. If you instead output a post-fader send, the reverb fades out in unison with dry audio.)



Pre- and post-fader effect and send routing for each track
***A** Input **B** EQ **C** Track volume **D** Track mute **E** Send **F** Effects Rack*

1. In the Sends area of the Mixer, click the send Power button.
2. Click the Pre-Fader/Post-Fader button to place the send either before track volume or after.
3. Set send Volume and Pan.
4. From the Send pop-up menu, select a bus.

For more information, see Insert effects before or after sends and EQ.

Equalize tracks

For each track, the Multitrack Editor provides a parametric equalizer.

? In the EQ area of the Editor or Mixer panel, do any of the following:

- Double-click the graph to access detailed controls in the Track EQ window. (See Parametric Equalizer effect.)
- Click the EQ Power button to compare audio with and without equalization.

Arrange and edit multitrack clips with Audition

When you insert an audio file in the Multitrack Editor, the file becomes a clip on the selected track. You can easily move clips to different tracks or timeline positions. You can also edit clips nondestructively, trimming their start and end points, crossfading them with other clips, and more.

To arrange clips in the Editor panel, you use the Move or Time Selection tools.

Select and move clips

? Do any of the following:

- To select an individual clip, click it in the Editor panel.
- To select all clips in selected tracks, choose Edit > Select > All Clips In Selected Track.
- To select all clips in a session, choose Edit >Select > Select All.
- To move selected clips, select the Move tool in the toolbar, and then drag the clips. Or choose Clip > Nudge Right or Nudge Left to move clips one pixel at a time. (If you zoom in to see individual samples, nudging moves clips one sample at a time.)

To move clips with the Time Selection tool *, right-click and drag (similar to the Hybrid tool technique in previous versions). You can also drag the clip header with any tool.*

Snap to clip endpoints

Snapping lets you quickly align clips with other clips. If snapping is enabled, both dragged clips and the current-time indicator snap to selected items. While you drag a clip, a white line appears in the Editor panel when snapping points meet.

1 To enable snapping for selected items, click the Toggle Snapping icon at the top of the Editor panel.

2 Choose Edit > Snapping > Snap To Clips.

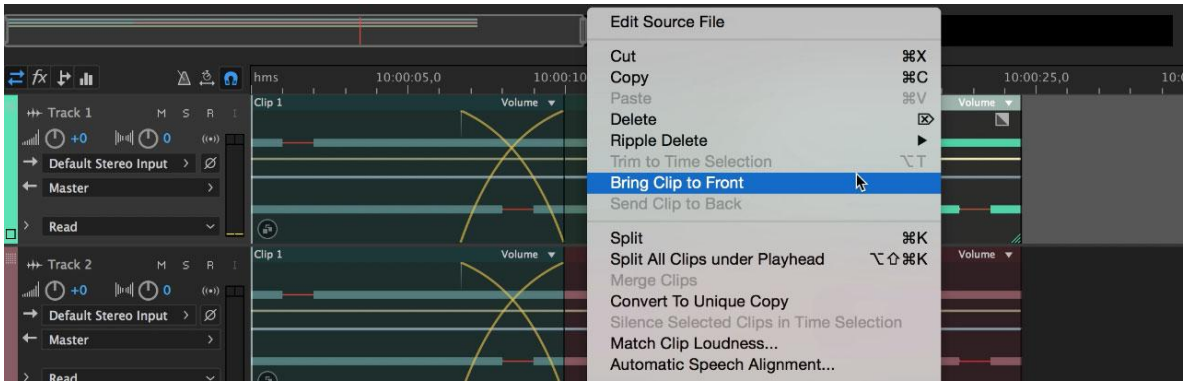
For more information, see Snap to loop beatsand Snap to markers, rulers, frames, and zero crossings.

Overlapping Clips

When clips overlap each other without crossfading, only the top-most clip plays.

You can change the order of the clips using any one of the following method:

• Select the **Bring Clip to Front** or **Send Clip to Back** command from the clip section of the main menu to rearrange the selected clip. • Select the **Bring Clip to Front** or **Send Clip to Back** command from the clip context menu to rearrange the clip. In case a clip overlaps other clips, the clips are listed in the **Bring Clip to Front** sub-menu where they are sorted by their start time to bring the hidden clips to the front.



Clip context menu

Copy a clip

You can create two types of copied audio clips: reference copies that share source files and unique copies that have independent source files. The type of copy you choose depends upon the amount of available disk space and the nature of destructive editing you plan to perform in the Waveform Editor.

Reference copies consume no additional disk space, letting you simultaneously edit all instances by editing the original source file. (For example, you can add the Flanger effect to a source file in the Waveform Editor and automatically apply the effect to all 30 referenced copies in a session.)

Unique copies have a separate audio file on disk, allowing for separate editing of each version in the Waveform Editor. (For example, you can add destructive effects to the version in an introduction while leaving the version in a verse dry.)

To quickly copy a reference, press Ctrl + C (Windows) or Cmd + C (Mac OS). Alternatively, Alt-drag (Windows) or Option-drag (Mac OS) the clip header.

1 Click the Move tool in the toolbar. Then right-click and drag the clip.

To copy with the Time Selection tool , right-click and drag the clip header (similar to the Hybrid tool technique in previous versions).

2 Release the mouse button, and choose one of the following from the pop-up menu:

- Copy Here (to copy a reference)
- Copy Unique Here

Trimming and extending clips

You can trim or extend audio clips to suit the needs of a mix. Because the Multitrack Editor is nondestructive, clip edits are impermanent; you can return to the original, unedited clip at any time. If you want to permanently edit an audio clip, however, you can quickly open the source file in the Waveform Editor. (See Comparing the Waveform and Multitrack editors.)

Remove a selected range from clips

1 In the toolbar, click the Time Selection tool .

2 Drag across one or more clips to select them and a range.

3 Do one of the following:

- To remove the range from clips and leave a gap in the timeline, choose Edit > Delete.
- To remove the range and collapse the gap in the timeline, choose Edit > Ripple Delete, and select one of the following options:

  Selected Clips Removes selected clips, shifting remaining clips on the same tracks.

  Time Selection in Selected Clips Removes the range from selected clips, splitting them if necessary.

  Time Selection in All Tracks Removes the range from all clips in the session.

  Time Selection in Selected Track Removes the range only from the currently highlighted track in the Editor panel.

Collapse a gap between clips on a track

? Right-click the empty area between the clips, and choose Ripple Delete > Gap.

Trim or extend clips

1 If you want to repeat a clip, right-click it and select Loop. (See Looping clips.)

2 In the Editor panel, position the cursor over the left or right edge of the clip. The edge-dragging icon appears.

3 Drag clip edges.

The **Clip > Trim** option has three parameters.

Trim to time selection If you want to use a portion of the clip, select the portion and choose Trim to time selection.

Trim start to playhead Place the playhead where you want your clip to start and choose this option to trim the clip.

Trim end to playhead Place the playhead at the position where you want the clip to end and choose this option to trim out the clip.

Shift the contents of a trimmed or looped clip

You can slip edit a trimmed or looped clip to shift its contents within clip edges.

Shift clip contents within clip edges

1 In the toolbar, click the Slip tool .

2 Drag across the clip.

Permanently edit a clip's source file in the Waveform Editor

? Double-click the clip header.

Split clips

Split audio clips to break them into separate clips that you can independently move or edit.

Split clips with the Razor tool

1 In the toolbar, hold down the Razor tool , and choose one of the following from the pop-up menu:

Razor Selected Clips Splits only clips you click.

Razor All Clips Splits all clips at the time point you click.

Tip: to switch between these modes in the Editor panel, press Shift.

2 In the Editor panel, click where you want the split to occur.

Split all clips at the current-time indicator

1 Position the current-time indicator where one or more audio clips exist.

2 Choose Clip > Split.

Set audio clip properties

In the Properties panel, you can quickly change multiple settings for selected audio clips. Clip settings for gain and mute are independent from similar track controls.

1 Select an audio clip, and choose Window > Properties.

You can access individual properties from the Clip menu.

2 Set the following options:

To change the clip name, type in the text box at the top of the panel.

Clip Color Click the swatch to customize. A swatch with a red slash indicates that the clip is using the default color for the current Appearance preset. (See Change interface colors, brightness, and performance.)

Clip Gain Compensates for a low or high volume clip that is difficult to mix.

Lock in Time Allows only up or down moves to other tracks, with a fixed timeline position. A lock icon appears on the clip.

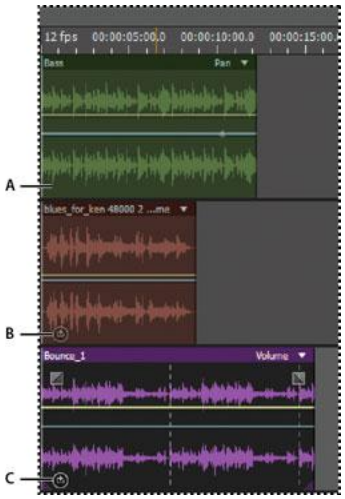
Loop Enables clip looping. For more information, see Looping clips.

Mute Silences the clip.

Looping clips

In many musical styles, you'll find loops used for everything from basic rhythm tracks to entire compositions. With Adobe Audition, you can either create your own loops or choose from thousands supplied for free via the Resource Central panel.

Loops let you create extremely flexible multitrack sessions. Though loops typically contain only one or two bars of music (four to eight beats), you can extend and repeat them by simply dragging with the mouse.



Identifying loops in the track display
***A** No loop **B** Single loop **C** Extended (repeated) loop*

Snap to loop beats

To better synchronize a loop-based session, use the Bars And Beats time format and enable snapping. Then add loops to create a rhythmic foundation, which you can build upon by recording new audio clips. (You can also add existing audio clips, but only their start or end points will align with loop beats.)

1 Choose View > Time Display > Bars And Beats. (This ruler format makes it easier to visually align loops with musical beats.)

2 From the Edit > Snapping submenu, choose any of the following:

Snap To Ruler (Coarse) Snaps to beats within bars. Use this option if you work with 1/4 or 1/2 bar loop files.

Snap To Clips Snaps to the start and end of audio clips.

Snap To Loops Snaps to the start and end of loops within clips.

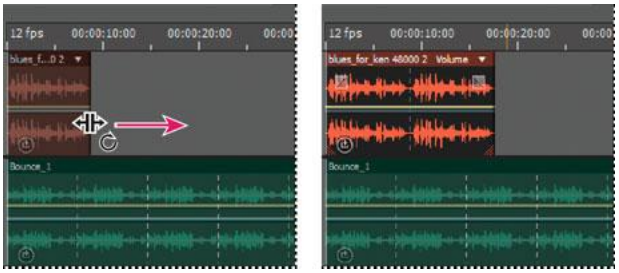
Enable a looped clip and change its length

1 In the Multitrack Editor, right-click an audio clip, and select Loop from the context menu.

2 Position the pointer over the left or right edge of the clip; the loop editing icon appears.

3 Drag to extend or shorten the loop.

Depending on how far you drag, you can make the loop repeat fully or partially. For example, you might drag a loop that is one bar long so that it extends 3-1/2 bars, ending on a beat within the loop. As you cross each bar, a white vertical line appears in the clip. This is the snap-to line, indicating perfect alignment to beats in other tracks.



Extending a loop

How to match, fade, and mix clip volume with Audition

To fade individual audio files, see Visually fading and changing amplitude. (The topics below address multitrack clips.)

Match multitrack clip volume

If multitrack clips have very different volume, making mixing difficult, you can match their volumes. Because the Multitrack Editor is nondestructive, this adjustment is completely reversible. To instead permanently change the volume of source files, see Match volume across multiple files.

1 Using the Move or Time Selection tool, Ctrl-click (Windows) or Command-click (Mac OS) to select multiple clips.

2 Choose Clip > Match Clip Volume.

3 From the pop-up menu, choose one of the following options:

Loudness Matches an average amplitude you specify.

Perceived Loudness Matches a perceived amplitude you specify, accounting for middle frequencies that the ear is most sensitive to. This option works well unless frequency emphasis varies greatly (for example, midrange frequencies are pronounced in a short passage, but bass frequencies are elsewhere).

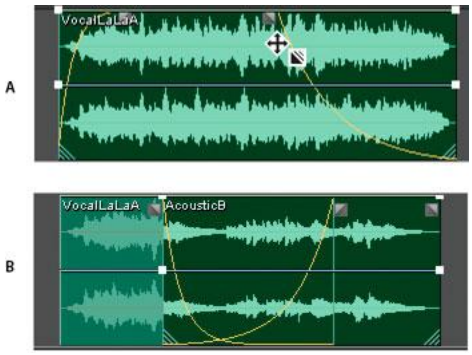
Peak Volume Matches a maximum amplitude you specify, normalizing the clips. Because this option retains dynamic range, it's a good choice for clips you plan to process further, or for highly dynamic audio like classical music.

Total RMS Amplitude Matches an overall root-mean-square amplitude you specify. For example, if the majority of two files is -50 dBFS, the total RMS values would reflect that, even if one file contains more loud passages.

4 Enter a Target Volume.

Fade or crossfade multitrack clips

On-clip fade and crossfade controls let you visually adjust fade curves and duration. Controls for fade ins and fade outs always appear in the upper-left and upper-right corners of clips. Controls for crossfades appear only when you overlap clips.



On-clip controls
***A** Drag controls in clip corners to fade in and out* ***B** Overlap clips to crossfade*

To

Fade a clip in or out

? In the upper-left or upper-right corner of the clip, drag the fade icon inward to determine fade length, and drag up or down to adjust the fade curve.

Crossfade overlapping clips

When you crossfade clips on the same track, you overlap them to determine the size of the transition region (the larger the overlapping area, the longer the transition).

1 Place two clips on the same track, and move them so they overlap. (See Select and move clips.)

2 At the top of the overlapping area, drag the left or right fade icon up or down to adjust the fade curves.

Fade options

To access the following fade options, select a clip, and then either right-click a fade icon in the Editor panel, or choose Clip > Fade In or Fade Out.

No Fade Deletes the fade or crossfade.

Fade In, Fade Out, or Crossfade If clips overlap, lets you choose the fade type.

Symmetrical or Asymetrical (crossfades only) Determines how the left and right fade curves interact when you drag them up and down. Symmetrical adjusts both fades identically, while asymetrical lets you adjust fades independently.

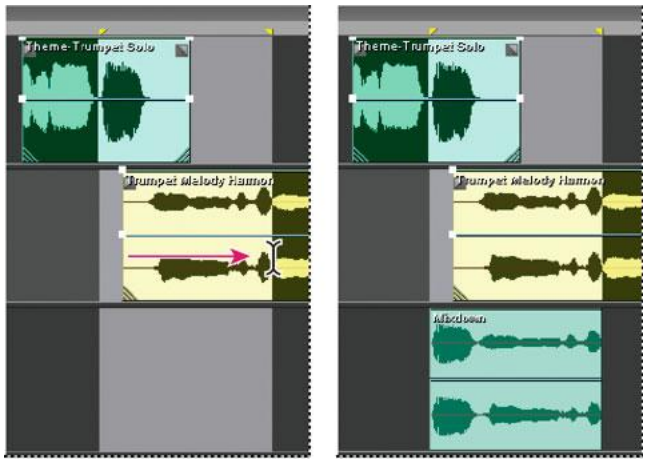
Linear or Cosine Applies either an even, linear fade or an *S*-shaped fade that starts slowly, then rapidly changes amplitude, and ends slowly.

***Tip**: To switch between Linear and Cosine modes while dragging fade icons, hold down Ctrl (Windows) or Command (Mac OS).*

Automatic Crossfades Enabled Crossfades overlapping clips. Deselect this option if automatic crossfades are undesirable or interfere with other tasks, such as trimming clips.

Create a single audio clip from multiple clips

You can combine the contents of multiple clips in the same time range, creating a single clip that you can quickly edit in either the Multitrack or Waveform Editor.



Creating single clip from multiple clips in Multitrack Editor

1 In the Editor panel, do any of the following:

- Select a specific time range. (See Select ranges in the Multitrack Editor.)
- Select specific clips if bouncing to a new track.
- Select nothing to mix down an entire session.

2 To combine the contents of the original clips, do either of the following:

- To create a track and clip in the Multitrack Editor, choose Multitrack > Bounce To New Track.
- To create a file in the Waveform Editor, choose Multitrack > Mixdown To New File.

To mixdown an entire session, see Export multitrack mixdown files.

Create an audio clip from a bus or master track

If you want to edit audio from a bus or master track, create a clip from the track.

1 In the Editor panel, select a time range for the new clip. (See Select ranges in the Multitrack Editor.)

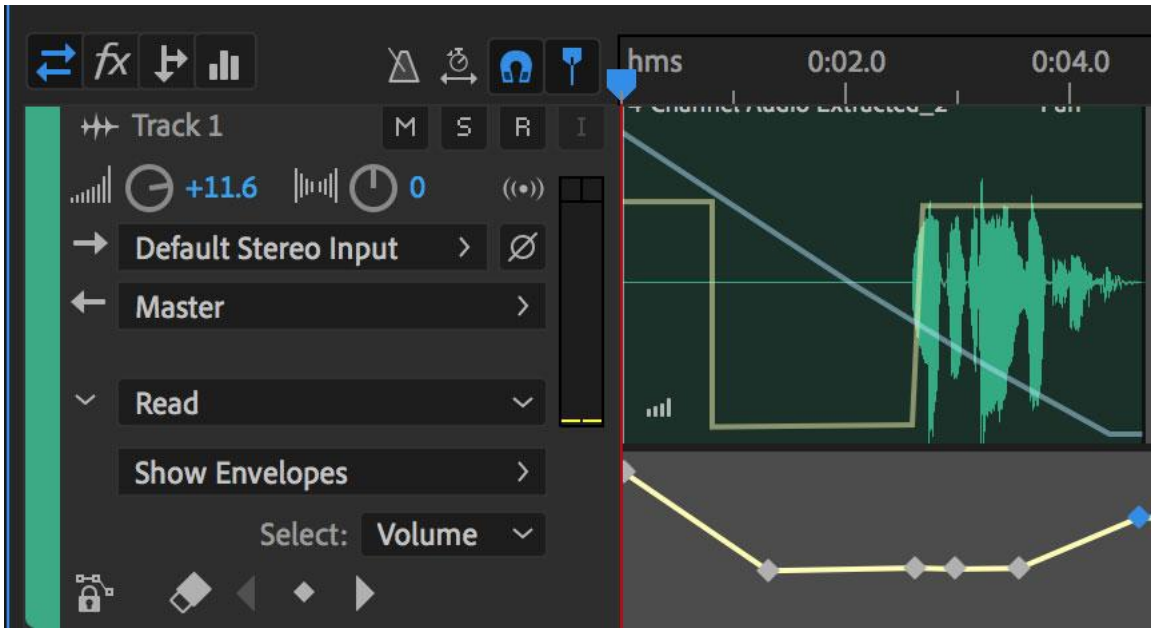
2 In the timeline, right-click a bus or master track, choose Bounce *[track type]* To New Track.

Adobe Audition creates a new track with a clip that reflects the bus or master mix.

Automating mixes with envelopes

By automating mixes, you can change mix settings over time. For example, you can automatically increase volume during a critical musical passage and later reduce the volume in a gradual fade out. See this video tutorial to watch mix automation in action.

Automation envelopes visually indicate settings at specific points in time, and you can edit them by dragging keyframes on envelope lines. Envelopes are nondestructive, so they don't change audio files in any way. If you open a file in the Waveform Editor, for example, you don't hear the effect of any envelopes applied in the Multitrack Editor.



Clip and track envelopes in the Editor panel

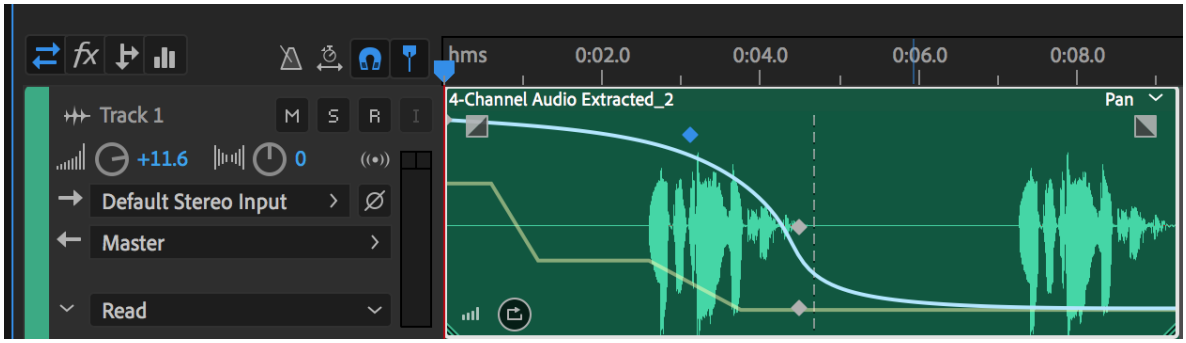
Automating clip settings

With clip envelopes, you can automate clip volume, pan, and effect settings.

On stereo tracks, clip volume and pan envelopes appear by default; you can identify them by color and initial position. Volume envelopes are yellow lines initially placed across the upper half of clips. Pan envelopes are blue lines initially placed in the center. (With pan envelopes, the top of a clip represents full left, while the bottom represents full right.)

Note:

On mono and 5.1 surround tracks, clips lack pan envelopes.



Two clip envelopes

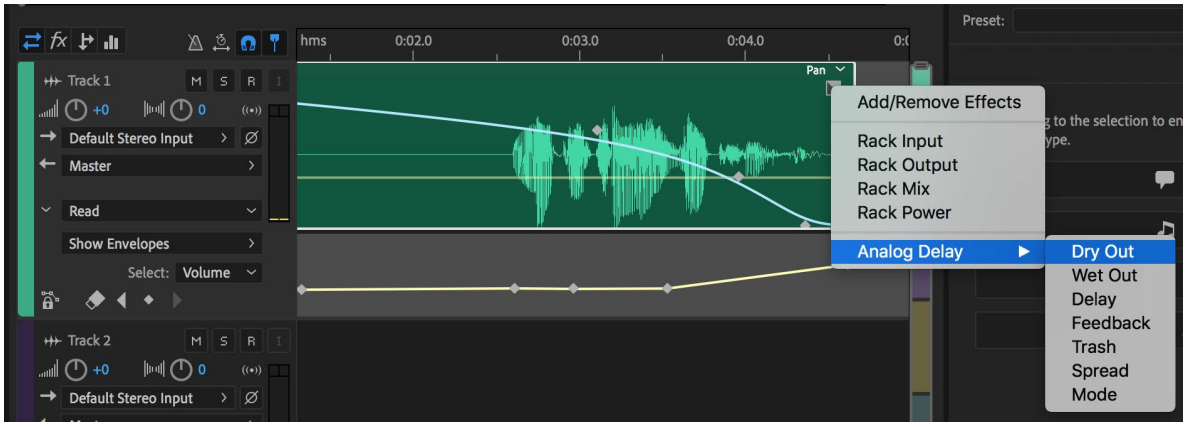
Show or hide clip envelopes

Clip envelopes are visible by default, but if they interfere with editing or are visually distracting, you can hide them.

? From the View menu, choose any of the following:

- Show Clip Volume Envelopes
- Show Clip Pan Envelopes
- Show Clip Effect Envelopes

Show or hide individual automation parameters



Showing a parameter for an effect

? In the upper-right corner of a clip, click the parameter menu, and select a Rack mixing option or an effect parameter. (In the menu, visible parameters appear with checkmarks. Select a parameter again to hide it.)

After you edit keyframes for a parameter, it appears with an asterisk (*) in the menu.

The Rack Power option lets you turn a clip's Effects Rack on and off over time.

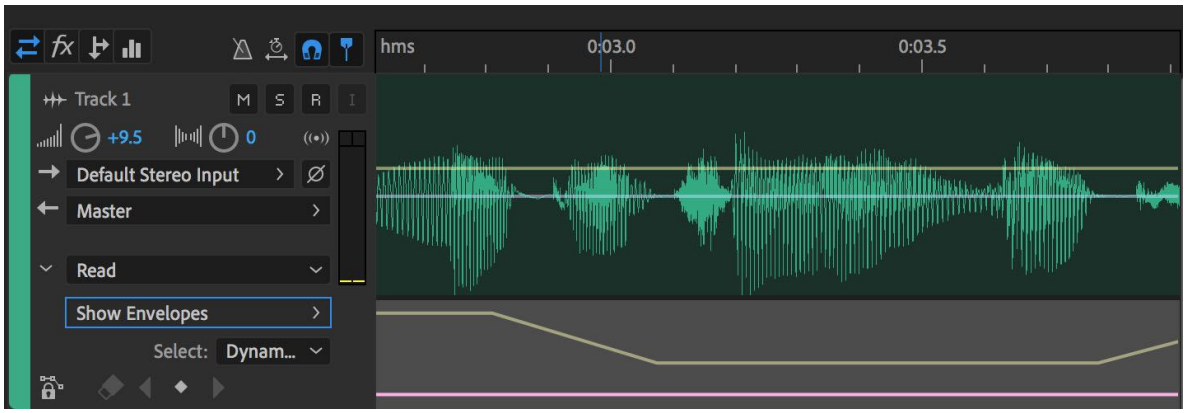
Disable clip keyframe editing

To avoid inadvertently creating or moving keyframes, disable keyframe editing.

? From the Multitrack menu, deselect Enable Clip Keyframe Editing.

Automating track settings

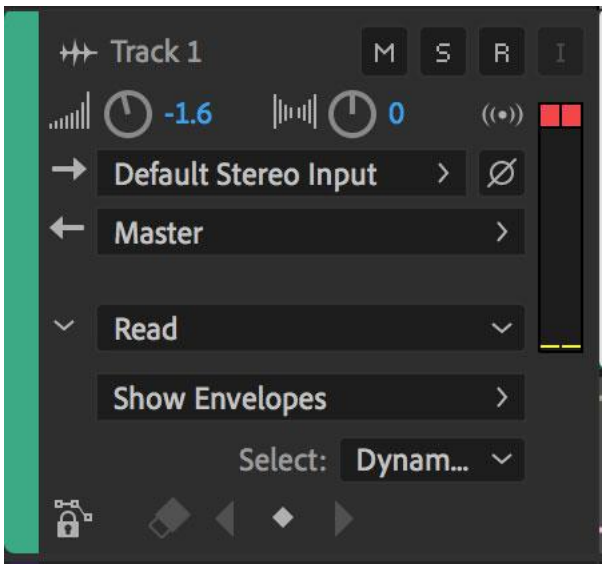
With track envelopes, you can change volume, pan, and effect settings over time. Adobe Audition displays track envelopes in an automation lane below each track. Each automated parameter has its own envelope, which you edit just like clip envelopes.



Automating track settings in the Editor panel

Create track envelopes

Track envelopes let you precisely change track settings at specific points in time.



Showing automation lanes in Editor panel

1. In Editor panel, click the triangle to the left of the Track Automation Mode menu for the track you want to automate. (The menu is set to Read by default.)
2. From the Show Envelopes menu, select a parameter to automate.
3. On the envelope line, click and drag to add and adjust keyframes.

Record track automation

While playing a session, you can record adjustments you make to track volume, pan, and effect settings, creating a mix that dynamically evolves over time. Adobe Audition automatically converts your adjustments into track envelopes, which you can edit with precision.

With an external controller such as the Mackie Control, you can adjust multiple settings simultaneously. See Control surface support.

1. In the Main panel, position the current-time indicator where you want to start recording automation.
2. Choose an option from the Track Automation Mode menu.
3. To start recording automation, start playback. As audio plays, adjust track or effect settings in the Editor, Mixer, or Effects Rack panels.
4. To stop recording automation, stop playback.

If recorded keyframes are too numerous or irregular, see Optimize recorded automation.

Track Automation Mode options

In the Editor panel or Mixer, you can choose one of the following modes for each track:

Off Ignores track envelopes during playback and mixdown, but continues to display envelopes so you can manually add or adjust keyframes.

Read Applies track envelopes during playback and mixdown, but doesn't record any changes you make to them. (You can preview such changes, but keyframes return to recorded settings.)

Write When playback starts, overwrites existing keyframes with current settings. Continues to record new settings until playback stops.

Latch Begins recording keyframes when you first adjust a setting, and continues to record new settings until playback stops.

Touch Similar to Latch, but gradually returns settings to previously recorded values when you stop adjusting them. Use Touch to overwrite specific sections of automation while leaving others intact.

Protect settings when recording automation

When recording automation, you may want to protect certain parameters from inadvertent changes, while adjusting other parameters.

1. At the left of the Editor panel, navigate to the specific track's controls.
2. Click the Show/Hide Automation Lanes icon [ICON].
3. From the Show Envelopes menu, choose the parameter.
4. Click the Protect The Parameter icon [ICON].

Optimize recorded automation keyframes

To prevent recorded automation from creating excessive or irregular keyframes, optimize the following settings in Multitrack section of the Preferences dialog box.

Automatch Time Determines how fast parameters return to original values in the Touch mode. You can set values from 0.00 to 5.00 seconds; the default value is 1.00 second.

Linear Edit Point Thinning Removes any keyframes that represent static, unchanging parameter settings.

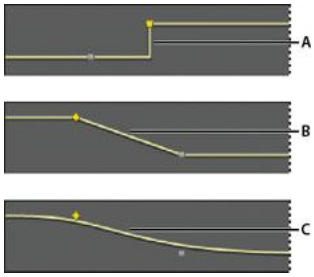
Minimum Time Interval Thinning Creates keyframes at time intervals larger than the specified value. Enter an interval between 1 and 2000 milliseconds in the Minimum Time field.

Adjust automation with keyframes

Keyframes on envelope lines change clip and track parameters over time. Adobe Audition automatically calculates, or *interpolates*, all the intermediate values between keyframes using one of two transition methods:

- *Hold* transitions create an abrupt change in value at each new keyframe.
- *Linear* transitions create a gradual, even change between keyframes.

You can also apply spline curves to an entire envelope, overriding the keyframe-specific setting above to create natural-sounding transitions that change in speed near keyframes. (See About spline curves for graphs.)



Transitions between keyframes
***A** Hold **B** Linear (the default) **C** Spline curves*

Add a keyframe

Do either of the following:

- Position the pointer over an envelope line. When a plus sign appears, click.
- Position the playhead where you'd like a track parameter to change. Then click the Add Keyframe icon in the track controls.

Navigate between track keyframes

1. In the Editor panel, choose a parameter from the Select menu near the bottom of the track controls.
2. Click the Previous Keyframe or Next Keyframe icon.

Select multiple keyframes for a parameter

- Right-click any keyframe, and choose Select All Keyframes.
- Hold down Ctrl (Windows) or Command (Mac OS), and click specific keyframes.
- Hold down Shift, and click to select a series of keyframes.

Reposition keyframes or the envelope line

- To reposition selected keyframes, drag them. (To maintain time position or parameter value, hold down Shift and drag.)
- To reposition a segment of an envelope without creating a keyframe, hold down Ctrl (Windows) or Command (Mac OS), and drag.

Change the transition between two keyframes

- Right-click the first keyframe, and select Hold Keyframe to abruptly change values, or deselect it to gradually transition from one value to the next.

Apply spline curves to an entire envelope

- Right-click an envelope line, and choose Spline Curves.

Delete keyframes

- Right-click an envelope line, and choose Delete Selected Keyframes. Or, drag an individual keyframe off a clip or track.

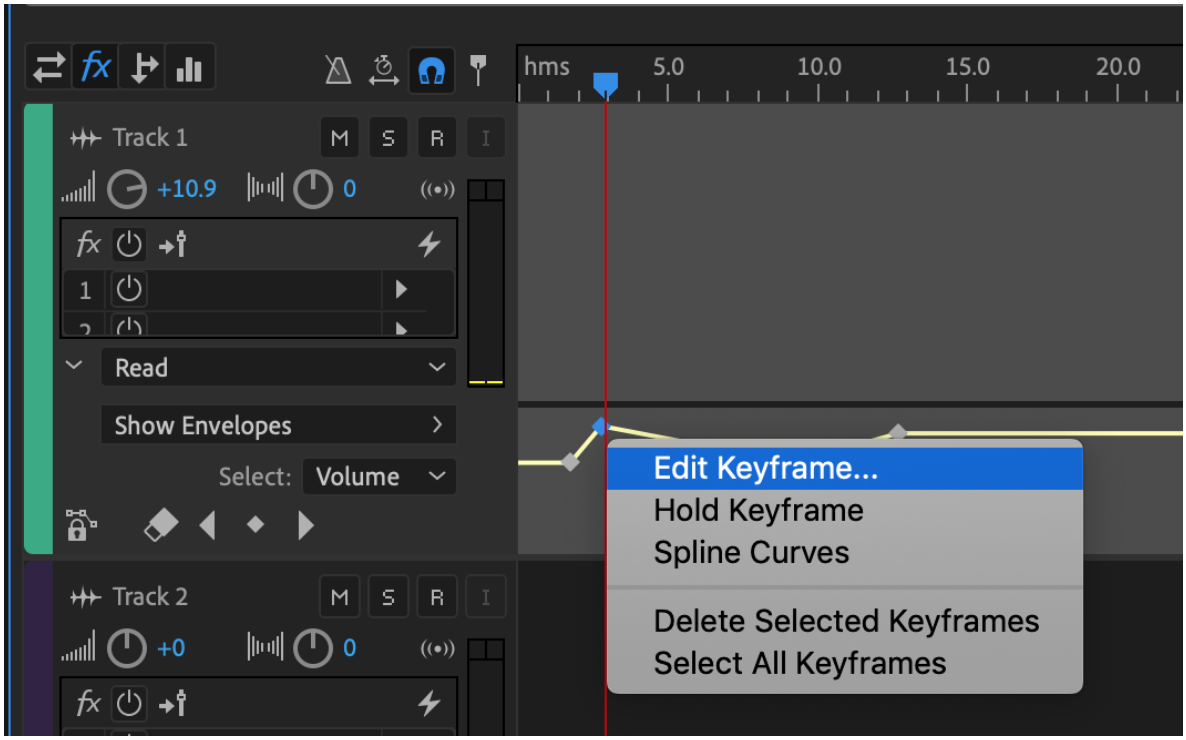
Disable clip keyframe editing

To avoid inadvertently creating or moving keyframes, disable keyframe editing.

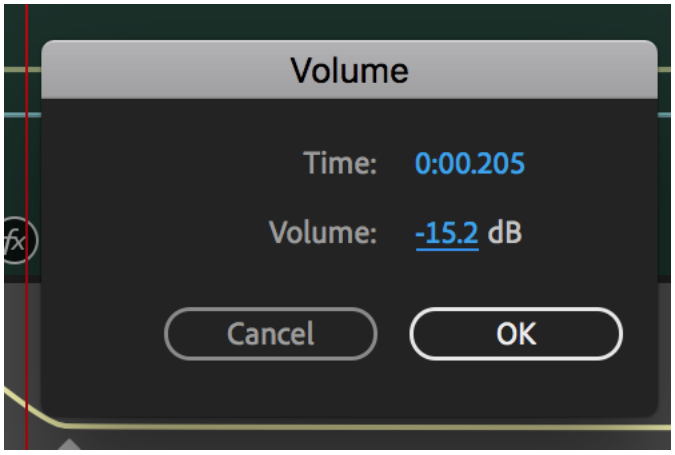
- From the Multitrack menu, deselect Enable Clip Keyframe Editing.

Edit keyframe manually

Right-click on a keyframe to open the context menu on a track automation envelope or a clip automation envelope in a Multitrack Session, and choose **Edit Keyframe**. A dialogue box will appear. Enter in a time value and a parameter value for the particular keyframe. For parameters which are not number-based such as on/off switches, there will be a tooltip on the second control in the dialog which shows the name for the parameter value.



Edit Keyframe



Enter parameter values

Hardware controllers and recordable automation

Mixing made perfect.

Hardware controllers and recordable automation

Multitrack clip stretching

1 Choose **Clip** > **Stretch** > **Enable Global Clip Stretching**.

2 In the **Editor** panel, drag the white triangles in the upper right or left corner of clips.

Select multiple clips to stretch them proportionately.

To customize stretch settings for selected clips, adjust the following **Stretch** options in the **Properties** panel:

Mode Choose from the following:

- **Off** disables stretching, reverting the clip to its original length.
- **Realtime** lets you hear the results of stretching as you drag clips. This mode is a good choice during the editing process.
- **Rendered (High Quality)** requires longer processing but can avoid audible artifacts. Choose this mode if you encounter slow playback or performance with Realtime mode.

Type Choose from the following:

- **Monophonic** is best for solo musical instruments or dialogue.
- **Polyphonic** is best for music with multiple instruments or complex ambient sounds.
- **Varispeed** changes pitch in addition to duration, similar to slowing down or speeding up analog tape machines.

*To quickly apply the same mode to all clips, choose **Clip** > **Stretch** > **Realtime All Stretched Clips** or **Render All Stretched Clips**.*

Duration, Stretch, and Pitch Provide numeric entries for more precise adjustments than dragging clips in the **Editor** panel allows.

Advanced settings Set the following:

- **Transient Sensitivity** (available in **Polyphonic** mode) Sets sensitivity to transients such as drum hits and note beginnings, which are used as anchor points for stretching. Increase if transients sound unnatural.
- **Window Size** Sets the size, in milliseconds, of each chunk of processed audio. Adjust only if echo or flanging artifacts occur.
- **Precision** settings (available in **Rendered** mode) determine the tradeoff between quality and processing speed.
- **Preserve Formants** (available when **Rendered** and **Monophonic** are selected) adjusts the timbre of instruments and voices, maintaining realism during pitch shifts.

To make male voices sound female, or vice versa, select Preserve Formants and dramatically shift pitch.

Chapter 8: Video and surround sound

Working with video applications

Edit audio clips from Adobe Premiere Pro or After Effects

From Adobe Premiere Pro® or After Effects®, you can quickly open selected clips in the Waveform Editor to restore or enhance them. When you save your changes, the updated audio automatically appears in your video projects.

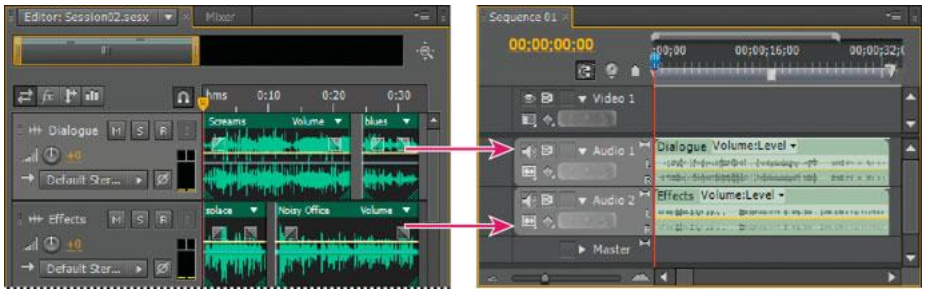
- In Adobe Premiere Pro, choose Edit > Edit in Adobe Audition > Clip.
- In After Effects, choose Edit > Edit in Adobe Audition.

Export a multitrack mix to Premiere Pro

Adobe Premiere Pro and Audition directly exchange audio between sequences and multitrack sessions. Any sequence markers appear in Audition, and separate tracks can be retained for maximum editing flexibility.

The applications exchange sequences and sessions using a shared XML format, exported to a folder that contains any referenced audio files.

If you want to send a single mixdown file to Premiere and don't need the option of exporting individual tracks, see Link sessions to exported mixdown files.



Multiple clips from Audition combined as stems in Premiere

1 Choose Multitrack > Export to Adobe Premiere Pro.

2 Specify a name and location for the exported session folder, and set the following options:

Sample Rate By default, reflects the sample rate of the original sequence. Select another rate to resample the file for different output mediums. (See Understanding sample rate.)

Export each track or bus as a stem Converts the full timeline duration of each track into a single clip, combining multiple clips if necessary. Select this option to extend and align clips with sequence start and end points.

Mixdown Session To Exports the session to a single mono, stereo, or 5.1 file.

Open in Adobe Premiere Pro Automatically opens the sequence in Premiere Pro. Deselect this option if you plan to edit the sequence later or transfer it to a different machine.

3 Click Export.

4 When Premiere Pro opens the exported XML file (either automatically or via the File > Import command), the Copy Adobe Audition Tracks dialog box appears.

From the Copy to Active Sequence menu, choose where the exported Audition tracks begin. Any new tracks are added below existing ones.

To send sequence audio from Premiere Pro to Audition, see Editing audio Adobe Audition.

Link sessions to exported mixdown files

To create layered video soundtracks that you can easily update, link multitrack sessions to exported mixdown files. When you select the exported files in an application such as Adobe Premiere Pro, you can either remix or edit them in Adobe Audition. As a video project evolves, simply repeat this process to create a polished final soundtrack.

Embed edit-original data in exported mixdown files

1 Open a multitrack session.

2 Choose Edit > Preferences > Markers & Metadata (Windows) or Audition > Preferences > Markers & Metadata (Mac OS).

3 Select Embed Edit Original Link Data in Multitrack Sessions.

4 When you export mixdowns, select Include Markers and Other Metadata.

Edit a mixdown file via a video application

1 In the video application, select a mixdown file you exported from Adobe Audition.

2 Choose Edit > Edit Original.

3 Select one of the following options, and then click OK:

- Open The Audition Multitrack Session That Created The File
- Open The File In The Audition Waveform Editor

4 Remix the linked session in the Multitrack Editor, or edit the mixdown file in the Waveform Editor.

5 To overwrite the original file in the video application, do one of the following:

- In the Multitrack Editor, choose File > Export > Multitrack Mixdown, and specify the same name and location as the original file.
- In the Waveform Editor, choose File > Save.

Importing video and working with video clips

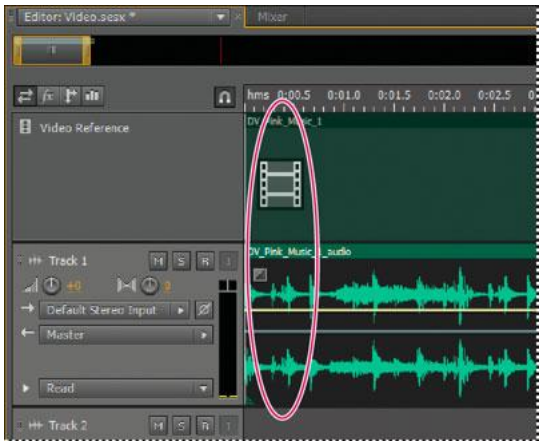
Insert a video file into a multitrack session

In the Multitrack Editor, you can insert video files to precisely synchronize a session with a video preview. When you insert a video file, its video clip appears at the top of the track display, and its audio clip appears on the track below.

You can move a video clip independently from the audio clip containing the original soundtrack. To instead keep such clips synchronized, before moving them, select both: Ctrl-click (Windows) or Command+click (Mac OS).

Note:

A session can contain only one video clip at a time.



Related video and audio clips (select both to keep them synchronized)

1. In the Multitrack Editor, position the current-time indicator at the desired insertion point.
2. Choose Multitrack > Insert File, and select a video file in a supported format. (See Video file formats.)
3. When you finish mixing audio for the video, export a mixdown, and import it into your video application. (See Export multitrack mixdown files.)

To quickly edit audio from a video file, choose File > Open. This technique is great for soundtrack editing that doesn't require a video preview, or for readapting soundtracks for audio-only mediums, such as radio or CD.

Snap to frames in a video clip

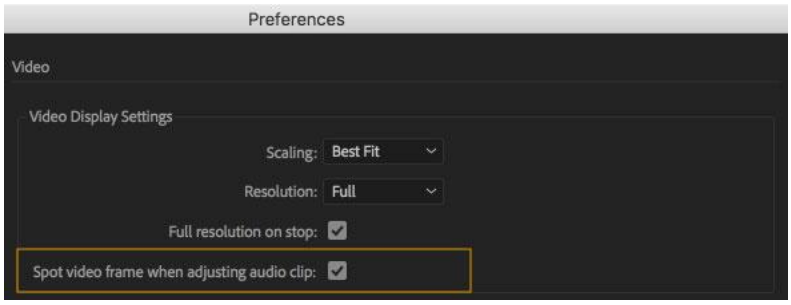
To synchronize a multitrack session with video, snap audio clips and the current-time indicator to frames.

1. Choose View > Time Display, and select the SMPTE time format that corresponds to the frame rate of the clip.
2. Choose Edit > Snapping > Snap To Frames.

Spot video frame preview while adjusting audio clip

While adding or moving audio clips in a timeline, the **Video** panel displays the frame underneath the play-head enabling you to perfectly align dialogue, sound effects, and musical crescendos. A blue dotted line on the timeline precisely indicates where the alignment spotting is occurring. To enable this feature, follow this step.

- **Preferences > Video > Spot video frame when adjusting audio clip.**



Video preference setting

Customize the Video panel

In the Video panel, you can preview video clips as a multitrack session plays to synchronize a soundtrack with specific video events. For example, scene changes, title sequences, or special effects.

To hide or show the Video panel, choose Window > Video.

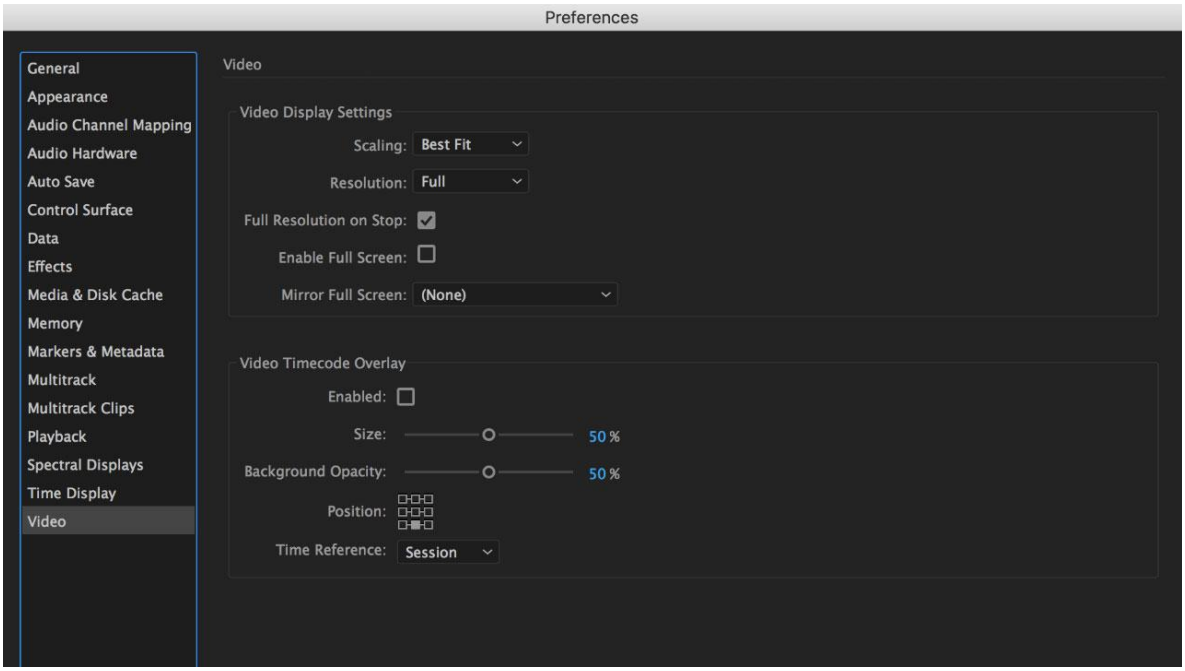
Video panel and related clip in Editor panel

To customize the Video panel, right-click it, and select either of the following options:**A zoom percentage**Zooms in or out.**Best Fit**Fits previews to the panel.

Timecode overlay

The Timecode Overlay displays the current time display string as an overlay on the video panel and the video fullscreen monitor. To set preferences for this feature, follow these steps:

1 Select **Adobe Audition CC > Preferences > Video** or right click the **Video Panel** and select **Video Preferences.**



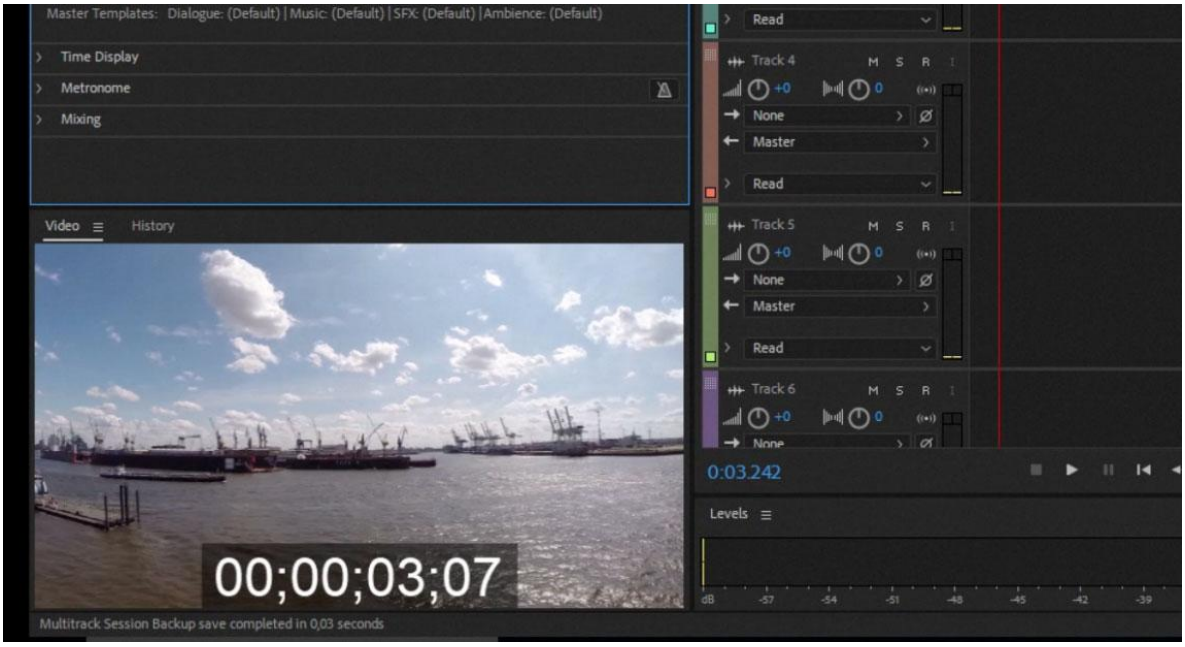
Timecode preferences

2 Use this window to set your preferences. You can adjust the following:

• **Enabled:** Enable or disable video timecode overlay. • **Size:** Adjust the size of the display of the timecode. • **Background Opacity:** Adjust the background opacity of the displayed timecode. •**Position:** Place the timecode as per your preference. • **Time Reference:** Choose if you want the time reference to be set to **Session** or **Media**.

Session: This option sets the start time of the multitrack session.**Media**: This option sets the internal time code of the media as preference.

The timecode overlay shows on the video panel once enabled.



Timecode overlay

Do more with Adobe Audition

Adobe Audition is a powerful application for recording and mixing audio, podcasting, and sound effect design.

5.1 surround sound

Monitoring 5.1 surround sound

Adobe Audition supports 5.1 surround sound, which requires five speakers, plus one low frequency subwoofer (LFE). To properly monitor 5.1 surround sound , your computer must have a sound card with at least six outputs, and those outputs must be mapped to the correct channels in Audition.

1 Choose Edit > Preferences > Audio Channel Mapping (Windows) or Audition > Preferences > Audio Channel Mapping (Mac OS).

2 Map each 5.1 channel to a sound card output:

- L: Front left speaker.
- R: Front right speaker.
- C: Front center speaker.
- LFE: Subwoofer
- Ls: Left surround speaker.
- Rs: Right surround speaker.

For more information, see Configure audio inputs and outputs.

Edit 5.1 surround files

In the Waveform Editor, you can edit 5.1 surround files with the same tools used for mono and stereo files.

To restrict editing to a subset of 5.1 channels, see Specify which channels you want to edit.

Pan tracks in a 5.1 surround mix

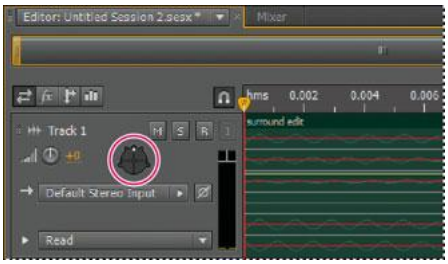
1 Open or create a 5.1 multitrack session. (See Create a new multitrack session.)

2 To open the Track Panner, do either of the following:

- Choose Window > Track Panner. Then select a mono or stereo track in the Editor panel.
- In the Editor or Mixer panel, double-click a surround plot for a mono or stereo track.

Note:

You can pan only mono and stereo tracks, not 5.1 tracks. To adjust the relative volume of channels in a 5.1 file, open it in the Waveform Editor.



Double-click a surround plot to open the Track Panner

3 In the Track Panner, do any of the following:

- To enable or disable channels, click the L, C, R, Ls, and Rs buttons. Or click LFE Only to send audio only to the subwoofer.
- In the large surround plot, drag to change the position of the signal.

  As you drag, white lines change in length from the speakers, reflecting the power of the signal in each. In the background, green and purple areas reflect the placement of left and right stereo image; blue areas indicate where the image overlaps.

- Below the surround plot, set the following options:

  AngleShows where in the surround field the sound appears to originate from. For example, -90° is directly to the left, while 90° is directly to the right.

  Stereo Spread Determines the separation between stereo audio tracks, with zero and -180° producing minimum separation, and -90° producing maximum separation.

  RadiusDetermines how far around the surround field the sound extends. For example, 100% produces a focused sound originating from very few speakers, while 0% produces an unfocused sound originating from all speakers.

  CenterFor tracks panned to the front of the surround field, determines the percentage of Center channel level relative to Left and Right level.

  LFEControls the level of signal sent to the subwoofer.

4 To pan additional tracks, simply select them in the Editor panel. The Track Panner automatically displays each track's unique settings.

To dynamically pan surround tracks over time, see Automating track settings.

Pan sends to a 5.1 bus track

1 In the Multitrack Editor, create a 5.1 bus track. (See Add or delete tracks.)

2 From the send output menu for an audio track, select the 5.1 bus track. (See Set up a send.)

In the Sends area of the Editor and Mixer panels, a surround plot appears. Double-click it to access the Track Panner for the send.

Chapter 9: Keyboard shortcuts

Default keyboard shortcuts

Keys for playing and zooming audio

These partial lists include the shortcuts that Adobe Audition experts find most useful. For a complete list of shortcuts, choose **Edit** > **Keyboard Shortcuts**.

Result	Windows shortcut	macOS shortcut
Toggle between Waveform and Multitrack Editor	8	8
Start and stop playback	Spacebar	Spacebar
Move current-time indicator to beginning of timeline	Home	Home
Move current-time indicator to end of timeline	End	End
Move current-time indicator to previous marker, clip, or selection edge	Ctrl+Left Arrow	Command+Left Arrow
Move current-time indicator to next marker, clip, or selection edge	Ctrl+Right Arrow	Command+Right Arrow
Toggle preference for Return CTI To Start Position On Stop	Shift+X	Shift+X
Zoom in horizontally	=	=
Zoom in vertically	Alt+=	Option+=
Zoom out horizontally	-	-
Zoom out vertically	Alt+minus sign	Option+minus sign
Add marker	M or * (asterisk)	M or * (asterisk)
Move to previous marker	Ctrl+Alt+Left Arrow	Cmd+Option+Left Arrow
Move to next marker	Ctrl+Alt+Right Arrow	Cmd+Option+Right Arrow

Keys for editing audio files

The following keyboard shortcuts apply only in the Waveform Editor.

Result	Windows shortcut	macOS shortcut
Repeat previous command (opening its dialog box and clicking OK)	Shift+R	Shift+R
Repeat previous command (opening its dialog box but not clicking OK)	Ctrl+R	Command+R
Open Convert Sample Type dialog box	Shift+T	Shift+T
Capture a noise reduction profile for the Noise Reduction effect	Shift+P	Shift+P
Activate left channel of a stereo file for editing	Up arrow	Up arrow
Activate right channel of a stereo file for editing	Down arrow	Down arrow
Make spectral display more logarithmic or linear	Ctrl+Alt+up or down arrow	Option+Command+up or down arrow
Make spectral display fully logarithmic or linear	Ctrl+Alt+Page Up or Down	Option+Command+Page Up or Down
Increase or decrease spectral resolution	Shift+Ctrl+up or down arrow	Shift+Command-up or down arrow

Keys for mixing multitrack sessions

Result	Windows shortcut	macOS shortcut
Select the same input or output for all audio tracks	Ctrl+shift-select	Command +Shift-select
Activate or deactivate Mute, Solo, Arm For Record, or Monitor Input in all tracks	Ctrl+shift-click	Command+Shift-click
Adjust knobs in large increments	Shift-drag	Shift-drag
Adjust knobs in small increments	Ctrl-drag	Command-drag
Nudge selected clip to the left	Alt+comma	Option+comma
Nudge selected clip to the right	Alt+period	Alt+period
Nudge selected clip up	Alt+up arrow	Option+up arrow
Nudge selected clip down	Alt+down arrow	Option+down arrow
Maintain keyframe time position or parameter value	Shift-drag	Shift-drag
Reposition envelope segment without creating keyframe	Ctrl-drag	Command-drag

Chapter 10: Saving and exporting

Save and export audio files

Save audio files

In the Waveform Editor, you can save audio files in a variety of common formats. The format you choose depends on how you plan to use the file. (See Audio format settings.) Keep in mind that each format stores unique information that might be discarded if you save a file in a different format.

1 In the Waveform Editor, do one of the following:

- To save changes in the current file, choose File > Save.
- To save changes under a different filename, choose File > Save As. Or choose File > Export > File to keep the current file open.
- To save currently selected audio as a new file, choose File > Save Selection As.
- To save all open files in their current formats, choose File > Save All.

Choose File > Save All Audio as Batch Process to process all open files. For details, see Batch process files.

2 Specify a filename and location, and choose a file format.

3 Set the following options:

Sample Type Indicates the sample rate and bit depth. To adjust these options, click Change. (See Convert the sample rate of a file.)

Format Settings Indicates data compression and storage modes; to adjust these, click Change. (See Audio format settings.)

Include Markers and Other Metadata Includes audio markers and information from the Metadata panel in saved files. (See Working with markers and Viewing and editing XMP metadata.)

Note:

Some CD-burning applications misinterpret non-audio information such as markers and metadata, producing an unpleasant burst of noise at the beginning of each track.

Export with Adobe Media Encoder

You can export your multitrack sessions to multiple formats by sending audio and video from Audition to Adobe Media Encoder. Adobe Media Encoder includes a series of formats and presets that allow you to render and publish high-quality audio and video output from Audition.

To export a multitrack session using Adobe Media Encoder:

1 Choose **File > Export > Export with Adobe Media Encoder.**

2 Specify a filename and location, and choose Adobe Media Encoder's file format as well as the preset.

3 By default, only the master track is exported with Adobe Media Encoder. To configure the routing of the tracks in your multitrack session to the channels in the output file, click **Change.**

4 If you want to export a specfic part of the project instead of always rendering the full timeline, use the **Range** option to choose **Time Slection**.

Note: To use the export workflow from Audition to Media Encoder, make sure that you are using the most recent updates of both the applications.

To learn more about using Adobe Media Encoder to render and export your output, see Encode video or audio items in Adobe Media Encoder

Extract audio channels to mono files

If you want to edit or output individual channels of a stereo or surround-sound file, extract them to mono files. Adobe Audition appends the channel name to each extracted filename. For example, a stereo source file named Jazz.aif produces mono files named Jazz_L and Jazz_R.

? In the Waveform Editor, choose Edit > Extract Channels To Mono Files.

Audition automatically opens each extracted file in the Editor panel.

Save multitrack sessions

A multitrack session file is a small, non-audio file. It merely stores information about locations of related audio files on your hard drive, the duration of each audio file within the session, the envelopes and effects applied to various tracks, and so forth. You can reopen a saved session file later to make further changes to the mix.

If you create multitrack mixes entirely in Adobe Audition, save session files in the native SESX format. If you plan to share multitrack compositions with other applications, however, see Export with Adobe Media Encoder.

You can save sessions with a record of all open files, and automatically reopen them with sessions. In the Multitrack preferences, select Reference All Open Audio And Video Files.

1 In the Multitrack Editor, do one of the following:

- To save changes to the current session file, choose File > Save
- To save changes under a different filename, choose File > Save As. Or choose File > Export > Session to keep the current session open.
- To save the session file and all the audio files it contains, choose File > Save All .

2 Specify a filename and location.

3 To include audio markers and information from the Metadata panel, select Include Markers And Other Metadata.

Note:

For a tutorial on exporting a multitrack session, see Exporting a multitrack session from Audition.

Export sessions to OMF or Final Cut Pro Interchange format

To transfer complete mixes to other applications in your workflow, export to OMF or Final Cut Pro Interchange format. OMF was originally created for Avid Pro Tools, but now is a common multitrack exchange format for many audio mixing applications. Final Cut Pro Interchange format is based on human-readable XML files, which you can edit offline to revise text references, mix settings, and so on.

Export to OMF

1. In the Multitrack Editor, choose File > Export > OMF.
2. Specify a filename and location.
3. Set the following options:

   Sample Type Click Change to access the Convert Sample Type dialog box. (See Converting sample types.)

   OMF Settings Click Change to access the following:

   Media Encapsulated stores audio clips in the OMF file itself for easier organization. Referenced stores audio clips in the same folder as the OMF file, letting you edit them offline if necessary.

   Note:

   Encapsulated OMF files are limited to 2 GB in size.

   Media Options Determines whether clip source files are trimmed to clip length in the Editor panel or reflect the entire original file.

   Handle Duration For trimmed clips, specifies the time duration to include beyond clip edges. Including additional audio provides more flexibility for fades and editing.

The Warnings section indicates elements of the session that will be excluded or changed. To copy this information to the clipboard, click the Copy Warnings button .

Export to Final Cut Pro Interchange format

1. In the Multitrack Editor, choose File > Export > FCP XML Interchange Format.
2. Specify a filename and location.
3. To include audio markers and information from the Metadata panel, select Include Markers And Other Metadata.
4. To combine session and source files for easy transfer to other systems, select Save Copies Of Associated Files. To change format and sample type for exported source files, click Options.

The following session elements are excluded from the exported file:

- The second channel of stereo clips. (All clips and tracks become mono.)
- Overlapping clips
- Effects and track EQ.
- Automation envelopes other than clip volume and mono-to-stereo track panning.
- Output routing, sends, buses, and the Master track.

Export session templates

Session templates include all multitrack properties and clips, helping you quickly start projects requiring similar settings and tasks. Examples include an intro theme or background ambience common to several related shows.

1. Choose File > Export > Session As Template. Then specify a name and location.

2 To apply a template to a new session, choose File > New > Multitrack Session. Then choose an option from the Template menu.

Export session archives

To combine session and source files for easy transfer to other computers or storage devices, export session archives.

Use this process to change the sample rate and bit depth of a session.

1 Choose File > Export > Session.

2 Select Save Copies Of Associated Files, and then click Options.

3 To save source files in a different format, select Convert Files, and set options as desired.

4 To export full or trimmed source files, choose from the Media Options menu.

Handle Duration specifies how much additional audio to include at the start and end of trimmed clips. For greater flexibility with future trimming and fading adjustments, add handles of up to 10 seconds.

Export multitrack mixdown files

After you finish mixing a session, you can export all or part of it in a variety of common formats. (See Audio file formats.) When you export, the resulting file reflects current volume, pan, and effects settings routed to the Master track.

To quickly mix specific audio clips down to a single track, use the Multitrack > Bounce To New Track command. (See Create a single audio clip from multiple clips.)

1 If you want to export part of a session, use the Time Selection tool to select the desired range.

2 Choose File > Export > Multitrack Mixdown and click one of the following depending on whether you want to export a selected portion of your audio, an entire session, or multiple clips:

- Time Selection
- Entire Session
- Selected Clips

(Alternatively, choose Multitrack > Mixdown Session To New File to open the mixdown in the Waveform Editor and skip the steps below.)

3 Specify a filename and choose a location and file format.

4 Set the following options:

Sample Type Indicates the sample rate and bit depth. To adjust these options, click Change. (See Convert the sample rate of a file.)

New Sample Type Indicates the resulting sample types of all the files in the panel after the conversion is applied.

Format Settings Indicates data compression and storage modes; to adjust these, click Change. (See Audio format settings.)

Mixdown Options Let you mixdown tracks as separate files, or simultaneously output mono, stereo, and 5.1 mixdowns. To adjust these and other settings, click Change.

In the Waveform Editor, choose Edit > Edit Original to open the multitrack session that created a mixdown file. This command requires embedded metadata in the file. (See Link sessions to exported mixdown files.)

Include markers and other metadata Includes audio markers and information from the Metadata panel in saved files. (See Working with markers and Viewing and editing XMP metadata.)

Open files after export Select the option if you want to open the files in Audition after conversion.

Audio format settings

In most cases, you should save uncompressed audio to the AIFF or WAV formats. Save to the compressed mp3 format only when creating files for the web or portable media players.

- In a Save As or Export dialog box, click Change to the right of Format Settings to access the options below.

To see the effect of settings changes, note the Estimated File Size at the bottom of the dialog box.

AIFF format settings (*aif, *.aiff, *.aifc)

AIFF is the standard uncompressed audio file format on Mac OS.

Sample Type (available only for 32-bit files) Sets the audio bit depth. Higher bit depths provide more dynamic range and reduce distortion, though they increase file size and processing time. To determine the proper setting for common output types, see Understanding bit depth.

Integer settings clip audio that extends beyond the amplitude range for a given bit depth. Floating Point settings require slightly more storage space, but retain much greater dynamic range.

Choose 32-bit Floating Point if you want to repeatedly process the file and discard minimal amplitude data.

Note:

To add Author metadata to an AIFF file, use the Dublin Core: Creator field on the XMP tab of the Metadata panel. (See Viewing and editing XMP metadata.)

Byte Order (available only for 16-bit files) Specifies the numerical sequence for bytes of data. Choose the Little-Endian method for systems with Intel processors and the Big-Endian method for systems with PowerPC processors.

Monkey's Audio format settings (.ape)

Monkey's Audio compresses files to roughly half their original size, using a lossless algorithm that retains sonic fidelity.

Compression Determines the tradeoff between processing speed and file size. (Fast processing produces larger file sizes, and vice versa.)

MP2 Audio format settings

MP2 (MPEG-1 Audio Layer II) is a lossy compression format particularly popular in radio broadcasting.

Channel Mode Choose one of the following:

- Automatic selects an option below based on the current channel configuation of the file.
- Mono produces a one-channel file.
- Dual Mono produces a two-channel file with identical audio in both channels.
- Stereo produces a two-channel file, processing channels independently, and retaining stereo imagery.
- Joint Stereo processes both channels together, reducing file size, but often affecting stereo imagery.

Bitrate Adjusts the tradeoff between file size and audio quality. Higher settings increase size and quality; lower settings reduce size and quality.

For information about Advanced options, position the mouse over them until a tool tip appears.

MP3 format settings

MP3 is the standard compressed audio format for online distribution.

Type Choose Constant to apply a consistent bitrate or Variable to change the bitrate based on audio content.

Bitrate (for constant bitrates) or Quality (for variable) Adjusts the tradeoff between file size and audio quality. Higher settings increase size and quality; lower settings reduce size and quality.

Avoid compressing the same audio to mp3 more than once. Opening and resaving an mp3 file causes it to be recompressed, so any artifacts from the compressing process become more pronounced.

QuickTime format settings (audio-only *.mov)

You cannot adjust format settings for audio-only QuickTime files, which are always saved as uncompressed PCM (Pulse Code Modulation) data.

Wave PCM format settings (.wav, .bwf)

Wave PCM is the standard uncompressed audio format on Windows.

You can include Broadcast Wave metadata in Windows PCM files. (See Viewing and editing XMP metadata.)

Sample Type (available only for 32-bit files) Sets the audio bit depth. Higher bit depths provide more dynamic range and reduce distortion, though they increase file size and processing time. To determine the proper setting for common output types, see Understanding bit depth.

Integer settings clip audio that extends beyond the amplitude range for a given bit depth. Floating Point settings require slightly more storage space, but retain much greater dynamic range.

Note:

The nonstandard 16.8 and 24.0 Floating Point options are provided for compatibility with previous versions of Adobe Audition. (16.8 was the internal format used by Audition 1.0. 24.0 was an optional setting from the same version.)

4GB Plus Support Stores files larger than 4GB using either RF64 format, which reflects the current European Broadcasting Union standard, or the Multiple Data Chunks format supported by older versions of Audition.

Note: To ensure compatibility with a wide range of applications, choose the RF64 format for files greater than 4GB.

libsndfile, FLAC, and OGG format settings

The following options are shared by a wide range of formats.

FLAC format uniquely is both compressed (reducing file size) and lossless (retaining full audio fidelity).

Format Specifies the file format and extension.

Encoding Specifies the data storage scheme for the file.

Byte Order Specifies the numerical sequence for bytes of data. The Default Byte Order automatically applies the default for your system processor and is typically the best option.

VBR Quality (OGG files only) Determines the tradeoff between file size and audio quality. (Higher settings produce larger files, but better quality.)

Close files

? Do any of the following:

- To close the current file in the Editor panel, choose File > Close.
- To close all open audio, video, and session files, choose File > Close All.
- To close files that aren't referenced by an open multitrack session, choose File > Close Unused Media.
- To close the current session and related audio clips in the Multitrack Editor, choose File > Close Session and Its Media.

Viewing and editing XMP metadata

In Adobe Audition, the Metadata panel provides BWF, ID3, and RIFF tabs that let you quickly access this common audio metadata. The XMP tab provides the same information, combined with a longer list of metadata common to Adobe video applications. (For example, the Display Title field on the RIFF tab corresponds to the Title field in the Dublin Core section of the XMP tab.)

Note:

To preserve metadata when you save files, select Include Markers and Other Metadata in the Save or Export dialog box.

The BWF tab applies to Broadcast Wave files, which let you specify a time offset for playback, as well as standard descriptive metadata. To include Broadcast Wave metadata in a file, you must save in WAV format; see Save audio files. To insert a Broadcast Wave file in the Multitrack Editor, see Spot-insert a Broadcast Wave file into a session.

The ID3 and RIFF tabs apply to mp3 and radio industry metadata, respectively.

About the Metadata panel and XMP

To streamline your workflow and organize your files, use *XMP metadata*. Metadata is a set of descriptive information about a file. Video and audio files automatically include basic metadata properties, such as date, duration, and file type. You can add details with properties such as location, director, copyright, and much more.

With the Metadata panel, you can share this information about assets throughout Adobe video and audio applications. Unlike conventional clip properties, which are limited to only one application's Project or Files panel, metadata properties are embedded in source files, so the data automatically appears in other applications. This sharing of metadata lets you quickly track and manage video assets as they move through your production workflow.

Note:

Properties in the Metadata panel also appear in Adobe Bridge, providing additional details that help you quickly browse assets.

About schemas and properties

A metadata schema is a collection of properties specific to a given workflow. The Dynamic Media schema, for example, includes properties like Scene and Shot Location that are ideal for digital video projects. Exif schemas, by contrast, include properties tailored to digital photography, like Exposure Time and Aperture Value. More general properties, like Date and Title, appear in the Dublin Core schema. To display different properties, see Show or hide XMP metadata.

For information about a specific schema and property, hover the pointer over it in the Metadata panel. For most items, a tool tip appears with details.

About the XMP standard

Adobe applications store metadata using the Extensible Metadata Platform (XMP). XMP is built on XML, which facilitates the exchange of metadata across a variety of applications and publishing workflows. Metadata in most other formats (such as Exif, GPS, and TIFF) automatically transfers to XMP so you can more easily view and manage it.

In most cases, XMP metadata is stored directly in source files. If a particular file format doesn't support XMP, however, metadata is stored in a separate *sidecar file*.

Project assets without corresponding files don't support XMP. Examples from **Adobe Premiere Pro** include Bars and Tone, Universal Counting Leader, Color Matte, Titles, Black Video, and Transparent Video.

To customize the creation and exchange of metadata, use the XMP Software Development Kit. For more information about XMP, see Extensible Metadata Platform.

Show or hide XMP metadata

To optimize the Metadata panel for your workflow, show or hide entire schemas or individual properties, displaying only those that you need.

1 From the options menu for the Metadata panel, select Metadata Display.

2 To show or hide schemas or properties, select or deselect them from the list.

Save, switch, or delete metadata sets

If you use multiple workflows, each requiring different sets of displayed metadata, you can save sets and switch between them.

1 From the options menu for the Metadata panel, select Metadata Display.

2 Do any of the following:

- To save a customized set of displayed metadata, click **Save Settings**. Then enter a name, and click OK.
- To display a previously saved set of metadata, select it from the menu.
- To delete a previously saved set of metadata, select it from the menu, and click Delete Settings.

Create schemas and properties

If you have a unique, customized workflow that the default metadata options don't address, create your own schemas and properties.

1 From the options menu for the Metadata panel, select Metadata Display.

2 Click New Schema, and enter a name.

3 In the list, click Add Property to the right of the schema name.

4 Enter a property name, and select one of the following for Type:

Integer Displays whole numbers that you drag or click to change.

Real Displays fractional numbers that you drag or click to change.

Text Displays a text box (for properties similar to Location).

Boolean Displays a check box (for On or Off properties).

Edit XMP metadata

Similarly named properties are linked in the Metadata and Files panels. However, the Metadata panel provides more extensive properties.

1 In the Editor panel, open the desired file or session.

2 In the Metadata panel, edit text or adjust values as needed.

Search XMP metadata

1 In the Editor panel, open the file or session you want to search.

2 In the Metadata panel, click the XMP tab.

3 In the search box, enter the text you want to find.

The list of metadata collapses to reveal only properties that contain your search string.

4 To navigate through the search results, click the Previous and Next buttons to the right of the search box, or press Tab.

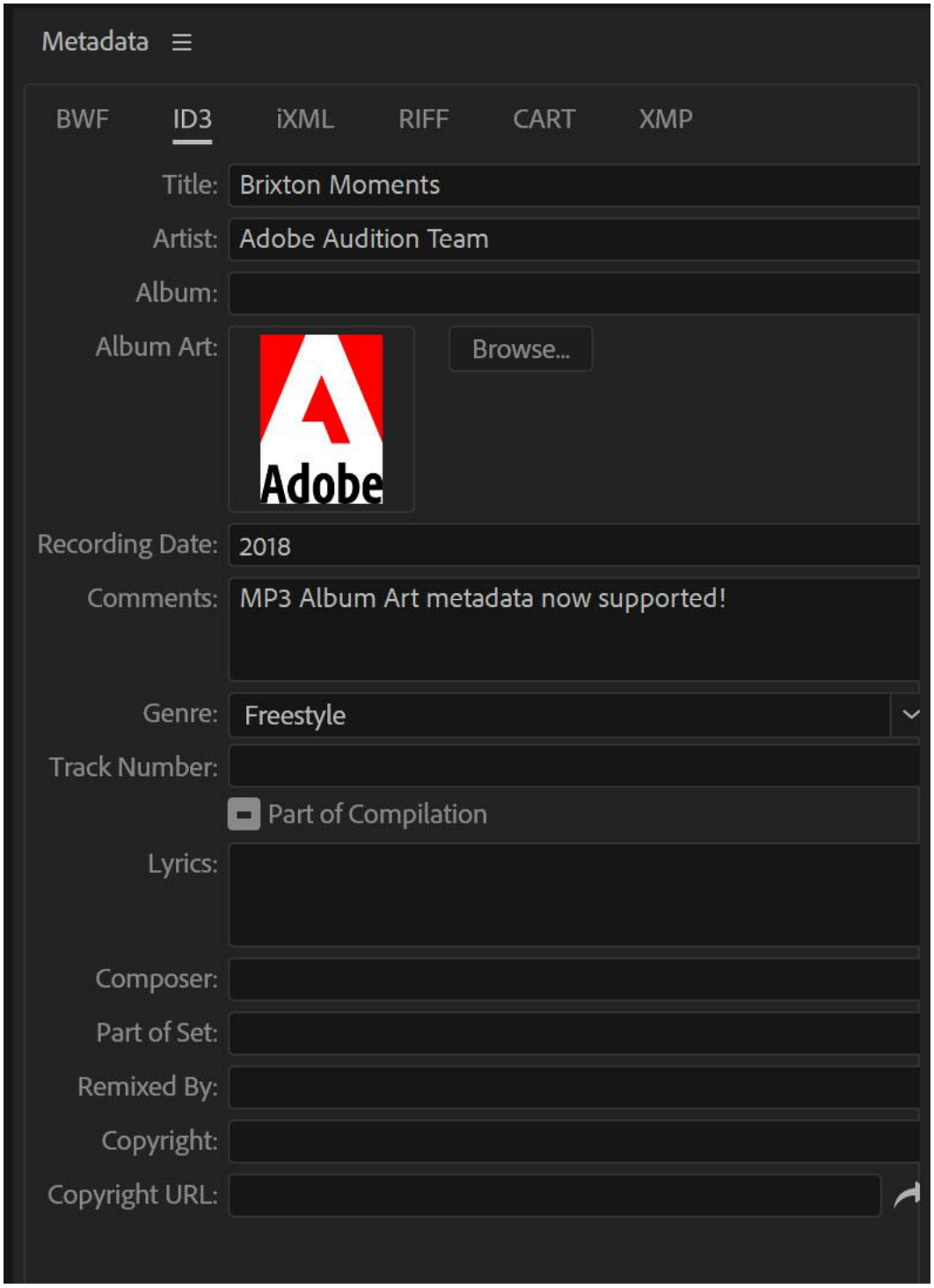
5 To exit the search mode and return to the full list of metadata, click the close button to the right of the search box.

Album Art metadata

Use the **Album Art** metadata option under the **ID3** tab on the **Metadata** window to choose the album art for your MP3 file.

To add album art, follow these steps:

1. To open the **Metadata** window, select **Windows > Metadata**.
2. Open the **ID3** tab from the list of available tabs.
3. Select **Album Art** option. The tab shows the first album art of the audio file by default.
4. To place a custom image as album art, use the **Browse** button. Note that only .png and .jpeg files are supported.



Album art for MP3 files

Note:

Album art works only with MP3 files. For other audio file formats, the album art entry in the ***Metadata*** *panel is grayed out.*